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TACTICAL VOICE COMMUNICATIONS OVER SHIPBOARD LOCAL AREA NETWORKS

by

Glenn R. Urie

December 2001

Thesis-Co Advisors: Luqi
John Osmundson

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The United States Navy's next generation ship(s) scheduled for commissioning in the year 2004 and beyond will integrate tactical shipboard voice communications systems into the local area network (LAN). A single network eliminates separate voice and data infrastructures, consolidates services, and reduces the cost of communications. The existing installation of high-speed shipboard data networks has laid the foundation for the convergence of these two technologies.

Currently, there is no high level definition of how multiple system types will share a common infrastructure. Neither is there a baseline defining acceptable end-to-end standards for the merger of these two systems. Common practice for demonstrating feasibility is confined to using commercial-off-the-shelf (COTS) equipment in a show-and-tell environment. Although this indicates certain operational features it does not demonstrate if telephony system's performance are within specified limits. Neither does this type of demonstration simulate realistic shipboard tactical load performance requirements or what effect this integration will have on data systems that co-habitat the LAN.

The purpose of this thesis is to define the convergence of the centralized shipboard tactical voice communication communications system into a distributed software-based system and the minimum set of acceptable software requirements for full integration of this system into the existing shipboard local area network infrastructure. In addition, this thesis will address the quality of service, tactical requirements risk assessment, interoperability, training, integration with legacy systems and other factors involved in the total cost of ownership.

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TACTICAL VOICE COMMINICATIONS OVER SHIPBOARD LOCAL AREA NETWORKS

Glenn R. Urie
A.S., Grossmont College, 1984
B.S., National University, 1987

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Author: ____________________
Glenn R. Urie

Approved by: ____________________
Luqi, Thesis-Co Advisor

John Osmundson, Thesis-Co Advisor

____
Luqi, Chair
Department of Software Engineering

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LIST OF SYMBOLS, ANCRONYMS AND/OR ABBREVIATIONS

1MC	General shipboard announcing system
μ	Micro (0.000001)
ATM	Asynchronous transfer mode
APBX	Automated private branch exchange
BRI	Basic rate interface
CBR	Constant bit rate
CCP	Call control Processor
COTS	Commercial off the shelf
COX	Central Office exchange
CSMA/CD	Carrier sense multiple access / collision detection
DS1	AT&T Bell System level 1 digital standard for systems operating at 1.544 Mbps and consisting of 24 DS-0 channels. Also referred to as T1.
GUI	Graphical user interface
FAX	Facsimile transmittal
FDDI	Fiber distribution data interface
E1	Abbreviation for the European equivalent of DS1 2.048 Mbps. European standard of T1
IADS	Integrated audio distribution system
IEEE	Institute of Electrical and Electronic Engineers
ISDN	Integrated services digital network
ISO	International standards organization
JB	Jack box
KITE	Keyswitched integrated terminal equipment
LAN	Local area network
MOS	Mean opinion score.
NOC	Network operations center
NT	Network terminal
OC1	Optical carrier level 1 (51.48 Mbps)
PA	Power amplifier
PBX	Private branch exchange
PABX	Private automated branch exchange
PC	Personal computer
POTS	Plain old telephone system
PPT	Push to talk
PRI	Primary rate interface
PSTN	Public switched telephone network.
QoS	Quality of service
SONET	Synchronous optical network
SPT	Sound powered telephone
STE	Secure terminal equipment
STU	Secure terminal unit
T1	AT&T Bell System level 1 digital transmission system operating at 1.544Mbps (1.536 Mbps excluding framing). Commonly used to refer to DS1.
TDM	Time division multiplexing
VoATM	Voice over asynchronous transfer mode
VoB	Voice over broadband
VoIP	Voice over internet protocol

I. INTRODUCTION

The Navy is making revolutionary changes in the way its forces exchange information to support military operations. Historically, the Navy has implemented a stovepipe methodology for shipboard communications systems. Each system is tailored to the individual requirements of its use. This encourages the installation of several similar technology systems with only slight differences in functionality.

The current naval communications infrastructure is stove-piped into several separate and distinct bandwidth communication channels. Voice, video, and data have employed specialized communications channels to proved features necessary for optimum communications. Many systems have been tailored to utilize these channels to provide specific capabilities and levels of service. For the integration of these requirements to be successful the infrastructure has to meet the total requirements set for each the individual deployed system.

Significant advances in the commercial world are allowing the implementation of combined technologies within a single environment. In the year 2004 the Navy plans to integrate shipboard communications of the three major communication components on a single channel on its first class of ship. Although LAN technology is advancing enough to support this effort, it has yet to reach the maturity level where all the required features for each communication type is available from a single technology source. In addition, the Navy has yet to identify requirements specifying capabilities and level of services required meeting shipboard functions.

To achieve network centric warfare objectives the Navy must overcome several obstacles. The Navy needs to leverage the advantage of commercial-off- the-shelf (COTS) systems while implementing features that are unique within the shipboard combat environment. The Navy's response to the challenge is to perform a high level examination of the requirements and select a solution from mutually exclusive commercial technologies. This provides a short term stopgap answer to the problem, but does not address the long term issues or requirements that are unique to the Navy.

The average lifecycle for a communication system is less than one year before it is eclipsed by technology that is faster with more enhanced features and less expensive. This creates a downturn in the Navy's ability to support a system in a cost-effective manner.

The challenge of the Navy is to integrate the requirements of the tactical voice communications systems into a single shipboard network. The integrated system will have to employ and support military forces more efficiently with reduced manning and budget requirements. It will have to support a robust infrastructure and information dissemination to dispersed forces while maintaining elements in achieving information superiority. It will also accelerate the transition to a microprocessor based tactical and support war fighting network. In general, by changing the tactical communications system the Navy will change the fundamental way in which it conducts information business.

II. SHIPBOARD TELEPHONY REQUIREMENTS

A. INTRODUCTION

Shipboard audio requirements have evolved from elementary forms of transmission and mechanical noise amplifiers that conveys the simplest forms of communication over a short distance. Because of the limitations of giving voice commands using these forms of communications devices, a system of distinct alarms was developed to communicate information over greater distance and adverse conditions. These alarms were distinct and could be heard above the noise of whatever activity was in progress at the time. These signals conveyed a more precise meaning of the sender intentions.

As shipboard current tactical voice communications systems have evolved the meaning and purpose of these alarms has remained a viable means of shipboard communications. Many of these features are unique to shipboard requirements and not available in standard commercial systems. For consideration as a tactical voice communications system, any future tactical voice system will have to carry these predefined alarms and perform several other shipboard requirements unique to the shipboard environment.

The system has to allow warfighters to exchange classified, unclassified, tactical and non-tactical information over the same consolidated infrastructure using the same end terminals. It has to have the ability, not only to push and pull essential information, but also to consolidate the information into a comprehensive tactical picture for the warfighter. This has to be seamless to the user in the field and tie even the smallest combatants together in a wide area tactical network. The end goal is to link all U.S. and allied forces' terminals into a fully integrated network that enables voice, video and data transmission network.

The system has to have a high degree of survivability both in environmental and operational longevity. The system is expected to function twenty-four hours a day, seven days a week for ten years without a failure. In the event a failure does occur, the system has to be able to perform self-diagnosis and identify the fault down to the lowest replaceable component then cut over to a hot standby backup unit. There can be no less than three non-related failures before the system shows any degradation in performance.

Several informal market studies have been conducted by several competing commands to present the Navy with a fully integrated voice communication system. There have been several promising results from these studies, but there has been no formal proposed solution. None of the proposed solutions is able to support of the total design requirements.

Because the military is not a significant market share in today's global economy it is difficult to find manufactures interested in making the required investment in developing a system specifically to meet the Navy's needs. Over ten to twenty years of new class ship construction the Navy would purchase a maximum of two hundred systems that would have to be maintained for another thirty-year life expectancy. This presents a limited window for the manufacture to recover their research and development fees. It also puts great demand on the government to maintain technology that is obsolete by the time it is installed.

The next best solution would be to have the Navy list the desired requires for a shipboard tactical system and have several commercial vendors compete to build these systems. This would allow those manufactures knowledgeable in the voice requirements to make minimal enhancements to their systems for shipboard use.

An alternate solution would be to have Navy communication engineers design and build a system that meets all the shipboard tactical requirements. This is a less desirable solution. Although, Navy engineers are familiar with the high level tactical requirements, it is much more difficult for the military to develop a skilled worker resource pool required to design, manufacture and maintain the system. However, this may be the only solution available to the Navy if it is unable to find a manufacture willing to assume the risk of development. If the Navy is to assume the risk of development then an in-depth understanding of the overriding factors effecting voice communications has to be undertaken.

B. EXISTING SHIPBOARD TELEPHONE SYSTEM

The tactical voice communications system is a shipboard audio frequency distribution system that is required to satisfy operation requirements for both tactical and administrative voice communications on twenty-four hour a day seven days a week basis. The current system provides computer centric controlled automatic switching service for line to line, line to net, and line to external connections. Sub-functions for subscriber services provide conferencing, multiparty nets, alternate addressing, abbreviated addressing, call forwarding, call transfer, and call override. The system needs to provide both open and secure communications capability. This ability will allow for the use of communication terminals, crypto equipment, radio interfaces and allow for direct connection of radio nets directly into the system.

The current tactical telephone infrastructure has been in existence for over thirty years and consists of equivalent of two private branch exchanges (PBX), one located in the forward part of the ship and the other located in the aft. The two systems are linked together via a proprietary digital basic rate interface (BRI) trunk line. The distribution of individual tactical phone units, network terminals (NT), are located throughout the ship in such a pattern that if a failure occurs in a unit the neighboring NT is connected to the alternate PBX. In the event of system failure, each PBX has hot backup features that allow it to automatically switch components without interruption to service. In addition, if one PBX fails or the interconnecting link fails the PBXs has the ability to operate independently. When the ship is tied up to the pier, each PBX can connect to a central office exchange (COX) allowing the PBXs to act as traditional administrative telephone system.

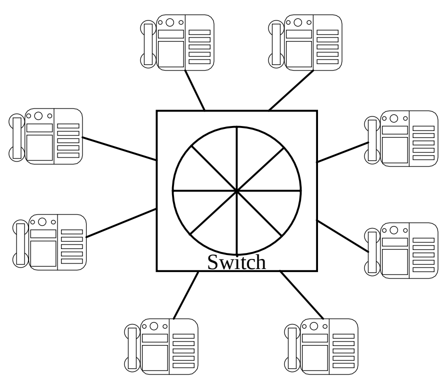

Figure 1. Time Slice Processing

The current shipboard tactical telephone system is a specialized centralized computer system that employs polling algorithms (Figure 1) where each NT is allocated a time slice within the polling circuit. The system operates by sampling the incoming signal at eight thousand times per second and converting each sample into an eight-bit value resulting in a sixty-four thousand bit data stream. Network terminals are locked into a common eight thousand hertz clock. Minor variations between the NT sample rate result in occasional clock slips, which are audible as clicks on the line. Poor data sampling within the switch poses difficulties to higher data rate electronic devices such as facsimiles secure voice devices that require a more precise sample rate for electronic data transfers.

Voice signals use two channels for each bi-directional connection that are multiplexed together between the PBX and terminal. During each time slice a sample is converted from the talking side of one phone and placed in the designated memory of the listen side of the receiver. This process is repeated for each terminal active on the switch at any time. If a terminal is not active, the processor completes a minimum amount of processing services then continues to the next terminal. There is no ability for the excess time of one slice to be reallocated into another time slice. At no time can the processing of a terminal exceed the maximum allowable time slice allocated to any given terminal.

The Navy's tactical voice system has failsafe requirements that are unique to the shipboard environment. The tactical voice system has to provide communications where a single point failure has virtually a zero probability of causing a catastrophic failure. Every major system and sub-system has a hot backup capability where switching to backup operations is automatic and causes no disruption in services. If catastrophic failure does occur, it has to happen in a prescribed manner and cannot affect operational components. If total system failure occurs, there has to be elements of the voice communication system that must remain operational.

There are analog and digital network terminal types connections into the tactical telephone system. Both types of network terminals have subcategories designed to meet specific needs of terminal interface types. All network terminal types connect into an interface that converts the format of the network terminal into the format of the telephony switch. This allows all network terminals, regardless of interface type, to communicate with each other through the tactical switch.

The system (Figure 2) has the ability to connect a minimum of two hundred tactical communication terminals (Table 1). One hundred are attached to each of the two independent interior communications switching centers (Table 2) and several of the full featured ISDN network terminals are attached to both centers, and are referred to as dual homed. The independent switching centers are functionally identical, capable of independent operation but mutually supporting. The subscribers are equally (approximately) divided between the systems. Each system contains five additional subsystems that provide connectivity for a specified number of terminals, networks, and trunks and interface connections.

Terminal Connection Type	Capability
ISDN Tactical Ports	48
ISDN Terminal Ports	72
Analog POTS NT Ports	24
Jack Box Nets NT Ports	60
SPT Nets NT Ports	8

Table 1. Terminal Interface Requirements

Trunk Connection Type	Capability
Inter ICSC Internodal Trunks	2
Shore Line Trunks / Channels (PRI)	6/144
Shore Line POTS Trunks (NT)	10
IADS Trunks NT Ports	10
PA / MC Trunks NT Ports	4
CCP / Ethernet Ports	2

Table 2. Trunkline Interface Requirements

Service blocking is the inability of the system to process requests because of non-availability of resources or the resources are currently be employed by other network terminals. Multi-user systems have, to some degree, an inherent form of blocking. If two users attempt to place a call at exactly the same moment, the system can handle only one request at a time. This forces the system to develop some sort of time-sharing or buffering that is a form of blocking. Because of the speed of the processor it may appear to the user that they are, in effect, the only user. This type of blocking is inconsequential and only blocking that impedes the system to the effect where it is recognizable or impacts the user should be considered.

The current tactical is defined as totally non-blocking. In effect the system has blocking because it can only handle sixteen active concurrent call attempts. All other service requests are placed in a queue and serviced in a first in first out (FIFO) buffering system basis. Non-blocking is implemented, in part, because of the load balancing and distribution of strategic terminals at critical locations that route calling internal to the switch and minimize calling that utilizes inter-switch connections. Only a limited number of terminals on each center have true non-blocking services. The remaining terminals, are in fact, impacted if these terminals are in use during critical functions.

Because of limited shipboard space requirements, the tactical voice communications must also function as an administrative telephone system when not employed in tactical mode. Additional services not normally associated with tactical requirements need to be included in the design and development of the new system. Services, other than those mentioned in as network terminals, including hotel services such as long distance tracking and logging, voice messaging, call waiting and call forwarding need to be incorporated.

The current system functionality and capacity has never been verified that it operates within specifications. When the system is used for administrative services a missed or blocked call is of minimal inconvenience. In tactical systems this is a high-risk method of system validation. The tactical communications aspect of the device is based exclusively on its longevity of service. Although there exists evidence of the survivability of the device under extreme physical conditions there is minimal testing or documentation to support the operational aspects under borderline or adverse usage. In loose terms: the equipment functions correctly because no one has been able to show that it does not function correctly.

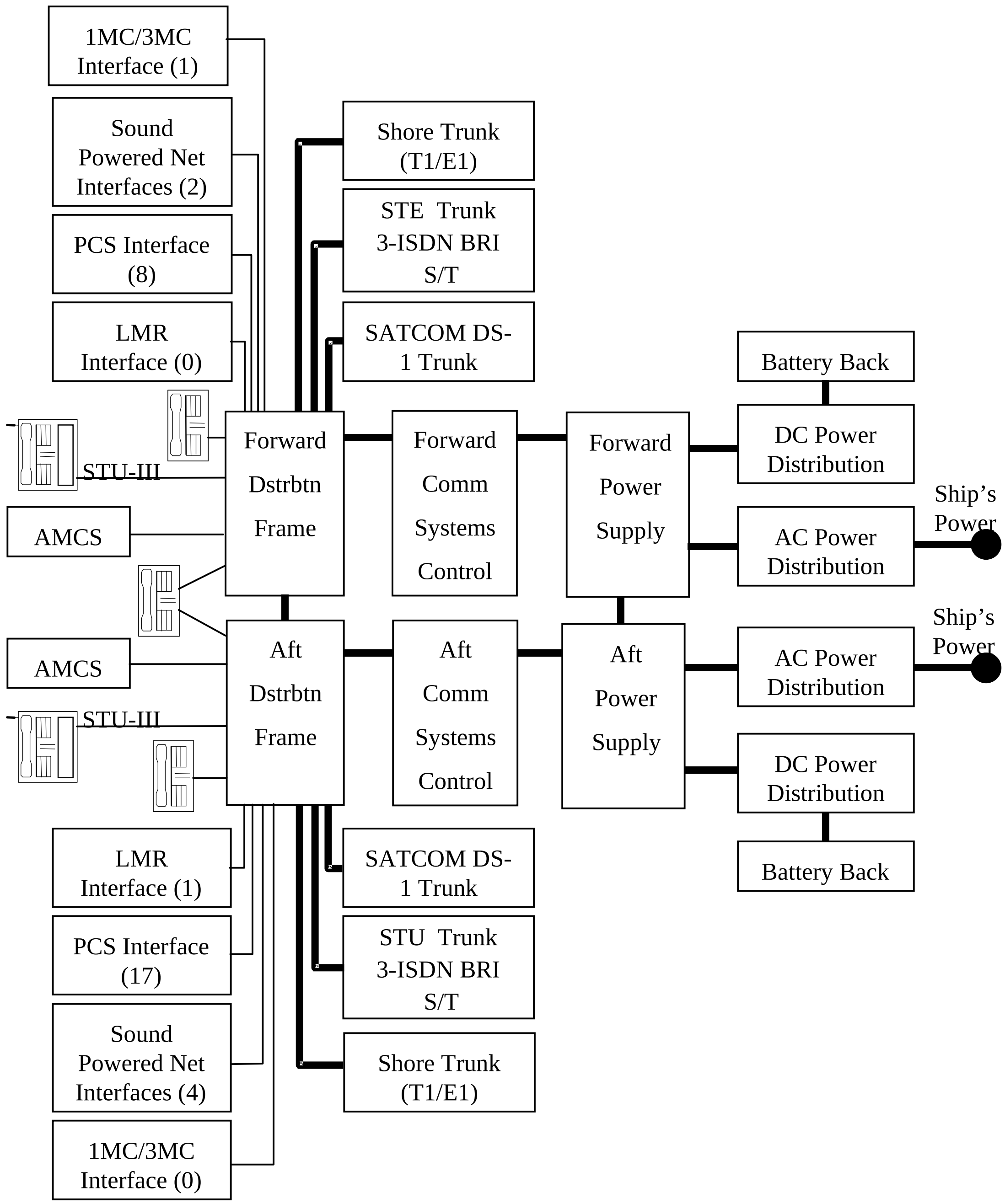

Figure 2. Tactical Switch Block Diagram

C. VOICE OVER THE SHIPBOARD NETWORK

Voice conversation over the network must include considerations from many different technical attributes that effect tonal quality of the final system. Many of these attributes are same as in traditional plain old telephone (POTS), but their causes and solution vary greatly as the system evolves into LAN based applications. Implementation of any portion of the LAN based telephone system needs to have a guaranteed service delivery rate across the entire LAN. The LAN needs to be capable of handling both voice and data traffic concurrently without any degradation of quality of service (QoS). Because tactical information will also co-reside on the LAN, a minimum set of service data requirements will exist.

The demand placed on the data network voice communications is very different from data transmission requirements (Table 3). Data traffic tends to be very bursty with widely varying amounts of required bandwidth and very tolerant of network delays. Voice traffic requires dedicated amounts of continuous bandwidth and is very sensitive to delay. Because current LAN environments are based on data transmission limitations, it will require determining the minimum best effort of the topography when deciding suitability of inclusion of specific protocols and formats. This requires far more resource sharing and places more demands on data network operations than the traditional best effort.

1. Elements Affecting Quality Of Service (QoS)

The ability of the system to precisely convert analog voice into digital format then accurately replicate it at the distant end is what determines the system's QoS. Analog phones are converted to digital format and operate by sampling the incoming voice signal at eight thousand samples per second. Each sample is converted into an eight-bit value generating sixty four-thousand bits per second data stream. This eight-bit information is then grouped and packaged with network data information and transmitted across the network. The information has to be received by the far end terminal in sufficient time for the network data to be removed and the information processed and converted back into analog form.

Qualities of service measurements are based on opinion sample polls where it is assumed that the mean opinion of the poll is more accurate than an evaluation based on technical merit. This means of obtaining technical evaluation is based on the assumption that the cross section of the sample is great enough to overcome any bias contained within the sample group. The rationalism for using an opinion poll is also based on the inability of the evaluation to utilize testing equipment with enough breadth and depth of ability to accurately measure the results of any load testing. Using this method of measuring quality of service presents a high risk environment of system validation for systems that are developed without the resources available to conduct large surveys. Within the Navy a new method of system validation needs to be developed that employs a method of system testing that can verify each of the interactive aspects that effect the quality of service.

Description	Data	Telephony
Switching Type	Packet	Circuit
Arrival Times	Bursty	Periodic
Quality Of Service	Best effort	Guaranteed
Delay Variation	Not Critical	Critical
Resource Allocation	Shared	Dedicated
Connection Type	Connectionless	Connection Oriented

Table 3. Data Versis Voice Transmission Criteria

The ability to send digital voice data over the LAN entails the encapsulation of voice data in progressive protocol formats. The protocols around the voice data are unwrapped as the data exits the format. On stand alone systems data formats can meet transmission criteria because there are no independent users. The shipboard LAN has competing users, many of whom are unknown to the tactical voice system. The flow of data across this unknown has to be controlled such that the sum total of all formats and relays of data across all portions of the system cannot exceed the criteria for any single quality of service element.

a. Latency

Latency is the delay, or the time it takes to travel, from the signal source to the signal destination through the circuit. In typical voice communications the listener acknowledges what the speaker is saying by giving verbal confirmation to the listener. In tactical communications these acknowledgment are incorporated into a formal command and affirmation language. If the system has significant delays, the natural response timing of the communication can be disruptive or confusing. In normal voice communications, latency becomes a problem when round trip ear-to-mouth travel time starts to exceed fifty milliseconds and become unacceptable when it exceeds two hundred milliseconds. In tactical communications latency should be kept to a minimum, at fifty milliseconds or less.

In plain old telephone (POTS) communication, where the circuit is established during communications, latency is primarily the result of distance traveled over the circuit. Because the current tactical telephone is enclosed within the confines of the ship, the delay resulting from distance traveled is minimal enough to be considered a non-factor. This is based on the assumption that the signal processor(s) has the ability to process each incoming signal at the minimum rate of once every one hundred twenty three microseconds (123µs).

In LAN based voice communications every step of the signal processing cycle introduces some sort of latency into the communications. Most of these steps introduce insignificant latency when no mitigating circumstances are considered. Latency becomes a problem when the system encounters more traffic than it can process and requests have to be buffered. In certain types of LANs latency can also be introduced as the number of requests for service increase. The latency of these systems is in direct proportion to the number of users vying for services within the system. In these conditions, a significant amount of latency can be introduced within a short distance. For the tactical voice system to be successful it has to be able to meet minimum system requirements under peak demand.

b. Jitter

Jitter is the variation in the arrival time of the signal from the sender. This problem is compounded in network communication because the communication is broken up into packets that are then sent across the LAN. Not all packets traverse the network following the same path or at the same rate. To remove jitter requires collecting packets and holding them long enough to allow the slowest packet to arrive in time to be played in the correct sequence. Latency is increased if the delay is too long.

In the standard PBX switch the analog signal for a network terminal is converted into digital format and held in the incoming buffer until the time slot of call control processor (CCP) processing the incoming signal. The digital signal is then processed and either placed into a service slot or in the outgoing buffer of the connected terminal. All timing is handled and assigned to the network terminal by the CCP. The speed and ability of the CCP to process incoming signals also determines the number of network terminals the system can process.

The CCP also has to handle call service requests that arrive at random times throughout the network terminal service cycle. These requests for service have to be processed within the time slot allocated to the designated network terminal. The CCP can only handle a finite number of requests before the processing time allocated to service all network terminals is exceeded. For low to average use, the ability of the CCP is able to process the service requests with non-noticeable delays. When the system is attempting to services requests during busy hour the tactical system becomes overloaded and queues any further requests until time can be allocated to the service.

In voice over LAN communications the requests for service to the telephony sever and the call procession from one telephone terminal sends the signal though several layers of asynchronous to synchronous signal conversions. Once the communication leaves the terminal there is little ability by the system to control the multiplex processing through the multiple layers. Each of these conversion layers has the possibility of introducing jitter into the communications. Under normal load conditions the system has the ability to meet load conditions with out significant introduction of jitter. As the system load conditions increase the system has to have the ability to meter and control transitions from asynchronous to synchronous protocols throughout the communications network.

To limit the effects of jitter and latency a clock signal has to be passed with the voice signal and service requests. The clock signal synchronizes all signals to other network terminals and all devices providing services to the system. Incorporating clock timing ensures that all signals are processed in a prompt manner and corrective action can be effected if signal processing is delayed beyond acceptable boundaries.

c. Clock Synchronization

Synchronization between the sending and receiving units is critical in maintaining overall QoS. Minor variations in clock frequencies result in clock slips, reducing the overall quality of service. If clocking is not properly synchronized it is most notable by an audible click. The most effective way to eliminate this problem is to inter-link all timing requirements into a single clock. The inter-linking of the clock is already established in synchronous communications requirements. Most data terminal equipment that currently exists for data LANs is asynchronous and not clock sensitive. This lack of synchronization limits most COTS end equipment as possible candidates for tactical communications systems.

In traditional PBX clock synchronization occurs within the call control processor. This centralized synchronization allows the system to process the maximum number of terminals for each predefined tick count. Voice over LAN utilizes not only parallel but also independent processing cycles. Before terminal to terminal communication is established clock cycles must be synchronized between communicating network terminals. Once the circuit is established the network terminals maintain QoS and the call server no longer is required to perform this service.

Not all telephony services need to be tied to the system clock. Call processing and progression tones from the telephony server and network terminals that are not voice communications can be non-interdependent. System commands, call setup, and takedown procedures do not need to be synchronized. These functions can be accomplished on a best effort tasking; as long as the maximum time utilized does not exceed the allocated time.

This is a major functional delineation between the traditional PBX and voice over LAN philosophies. Traditional PBX is a time allocated process where the service provided to the time slot of the network terminal has to provide all services requested by that NT or be placed in a queue. This functional requirement also has to be provided for those terminals that are on-hook and not currently being serviced by the system. The total number of allocated time slots can not exceed the total number of terminals connected to the system.

Voice over LAN is a load sharing strategy where there are loosely coupled multiple processes' handling exactly the same tasking. When the terminal request service the processors handling that service determine among themselves which processor is least loaded and that processor services the request. Every processor is aware of the completed functions all processors within the domain of the system. If a processor fails in mid function that function is lost and the terminal operator is required to re-request the desired function. Because of the independent nature of the system no completed function of a terminal is lost due to loss of a telephony service processor or network component. Each requests for service is independent of any previous service request.

d. Talker Overlap

Talker overlap is the perceived silence of the speaker and the listener talking when the speaker did talk and before the communication reached the listener or when one talker steps on the speech of another. Talker overlap is a unique form of latency. The signal appears to be functioning normally with the exception that there is enough delay in the signal processing to replicate the effect of long distance propagation. This problem becomes significant when the one-way transmission delay is greater than two-hundred milliseconds.

The primary cause of talker overlap in voice over LAN is the result of the asynchronous digital signal not being reassembled into synchronous format in a timely manner. The vehicle employed to transfer voice from the terminal to the LAN needs to service a timing or polled sequencing that ensures a minimum level of service to each time sensitive device (Table 4). The most common data network employed by the Navy today is a best effort service and does not employ any mechanism for ensuring a minimum level of effort.

Accumulation Delay	Latency In Milliseconds
Processing Time	10
Framing Time	30
Buffering	0
Packetizing	30 (two frames per packet)
Jitter	30
Media Access Delay	10 (5 – 2 msec hops)

Table 4. Average Latency Summary Times

e. Voice Compression

Voice compression is the ability of the signal to be reduced in size without compromising voice clarity or increasing latency. Voice compression would allow a greater number of concurrent calls utilizing a lower bandwidth of the LAN. Voice compression is of minimal value in an enclosed shipboard LAN tactical PBX and should only be considered if the overall development is approaching bandwidth limitations of the network backbone.

Unlike signals that are generated in native digital format, that can be compressed and decompressed with no virtual loss is signal quality, every point along the curve of the analog voice signal is important and any compression of the active signal causes some loss in quality of service. For the human ear, minor losses are not a critical factor. For mechanical methods of transmission, such as modem or FAX machines, this is such a significant loss in signal quality that the signal can not be reproduced with sufficient fidelity at the receiving end to be of usable value.

Even though voice communications is a relatively slow changing medium almost all compression algorithms have a certain loss in signal fidelity. Loss of signal quality is directly related to the degree the signal is compressed. In true voice communications a significant amount of signal degradation can occur before the signal becomes unintelligible to the human ear. Use of the voice range of frequencies signal by mechanical devices employing phase lock loops or other synchronous communication practices are not as forgiving in their signaling requirements. Even partial distortion of part of a waveform can cause signaling malfunctions.

The Nyquest solution of have at least two points on the curve to represent any frequency within the analog signal should be used as the minimum solution for tactical voice communications. The means the frequency represented in the voice spectrum at a clock sample rate of eight kilohertz analog to digital conversion is four thousand hertz. Any data compression has to maintain this integrity or the quality of the signal is degraded enough that it is not considered reliable.

There are two current standards for interoperability of telephony signal compression. The utilization of data compression is not implemented across all telephony signals. This lack of common signaling syntax creates challenges when attempting to interconnect equipment or PBXs from different manufacturers. This requires the signal to be expanded and compressed for each device that is not equipped with the signaling ability to handle the compressed format.

If voice compression is employed, it needs to be implemented in the network terminals as part of the audio communications and not part of the call signaling protocols. This allows the calling terminal to determine the abilities of the receiving terminal to handle predetermined compression algorithms and to establish communications based on those abilities during the call setup procedures. If the calling terminal does not request or the receiving terminal does not respond to the request for compression protocols then non-compression protocols should be employed.

f. Silence Suppression

Voice communications uses a bi-directional channel. Typically, only one person is talking at a time; thus one channel goes unused. Silence suppression can save up to fifty-percent on the bandwidth by suppressing the non-used channel. This requires more sophistication for the network gateway. For effective bandwidth utilization the system should apply silence suppression to each voice data stream as required.

This creates unique demand on the network monitoring and control capabilities. As the voice is suppressed the channel is released and the bandwidth becomes available for reallocation. When the voice threshold is exceeded the channel is re-established and the bandwidth availability is reduced. Although this feature becomes extremely useful when considering hundreds of thousands of on demand circuits and the relative available bandwidth, it becomes impractical to implement for shipboard use in the first phases of development because of the difficulty of implementation and the virtual unlimited bandwidth availability on the ship’s data backbone.

g. Echo Suppression

An echo is created by the reflection of the speaker’s voice from the far end equipment back to the speaker’s ear. The echo is a human comfort factor that gives the speaker feeling that there communication is being transmitted to the far end terminal. Echo becomes a problem when the round trip signal takes longer than fifty milliseconds.

The fifty-millisecond round trip echo suppression also limits the total latency allowed within the system.

If the latency is greater than fifty milliseconds then the system implements an echo suppression algorithm. The echo suppression firmware needs to be located within the network terminal equipment. As the signal is transmitted a timing mark is added to the digital signal. If the receiving terminal receives the incoming signal with a time delay of less than twenty-five milliseconds the part of that signal is picked up by the far end terminal and retransmitted back to the receiving terminal. Care has to be taken to ensure the echo gain is minimal and not sufficient enough to cause system feedback. If the delay at the far end terminal is greater than twenty-five milliseconds the transmitted signal is not combined and retransmitted at the far end terminal. The far end terminal does inject a low level background noise to give the speaker a physiological effect of bi-directional communications.

2. Mean Opinion Score (MOS)

The mean opinion score (Table 5) is an industry employed method of determining the effectiveness of the overall quality of the signal through the PBX switch. On the mean opinion score (MOS) scale, a zero equals the worst quality of sound quality and a five is the highest. MOS is a subjective means of determining the effectiveness of the transmission and is designed to be a reflection the mean opinion of the listeners' for the system being presented.

MOS Score	Quality Of Service
4.0 to 5.0	Toll or Paid Service
3.0 to 4.0	Local Service
Below 3.0	Not Acceptable

Table 5. Quality Of Service Opinion Ratings

This is method of determining signal quality is arbitrary in nature and it is difficult to derive a baseline or comparison of the collective signal properties with a limited sample set. It is difficult for the Navy to employ a sample population large enough to create an accurate determination of the actual signal quality. It is also very costly to repeat the evaluation process for any subsequent changes or upgrades within the life cycle maintenance of the system. Because of the arbitrary nature in determining MOS, it is recommended that the Navy approach the evaluation of its signal quality adopting a more deterministic method. This would allow independent development of the LAN based tactical BPX while the Navy should perform system evaluations to determine the degree of performance on compliance to the desired standards.

Even in the best case scenarios there is a certain amount of unavoidable signal quality loss. The sample rate in the digital conversion and the inability, even in digital format, to replicate those signals at the distance terminal introduces loss of signal quality (Figure 3). This ability of the distant end terminal to replicate the signal between sample points is based on extrapolation algorithms and may not accurately represent the actual signal transmitted. This is based on the analog to digital sample rate and is determined by the means percentage of accuracy between the test signal at the transmitting terminal and the accuracy of reproduction of the sample at the receiving terminal.

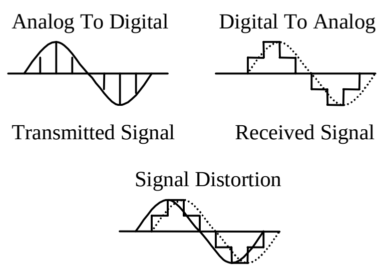

Figure 3. Signal Quality

The ability to measure under no load conditions is relatively simple and straightforward and can be automated without significant effort. It is crucial that the tactical system be measured under both no load and busy hour conditions. The busy hour or maximum load testing is not a simple procedure and can be costly in time and effort. Testing under busy hour requires the ability of the tester to have precise control of all aspects of the entire system and to know precisely at what point in the cycle each NT is functioning. The testing system has to be able to emulate multiple call conditions on hundreds of terminals and track the timing and circuit conditions of the tactical switch during each period of the test. For this reason the Navy has opted to forgo load testing on its pervious tactical telephone systems.

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III. TELEPHONY BASED TACTICAL SHIPBOARD COMMUNICATION SYSTEMS

A. TACTICAL TELEPHONY

The primary function of any voice communication system is to provide a means of connecting communications services from one communications network terminal (NT) to another. This is further expanded in modern voice communications by regulating additional services, depending on the class a service and classmarks. This service allows subscribers to place and receive multiple concurrent calls by providing each network terminal functionality to adequately service all calls. Tactical terminals must provide non-blocking services allowing as many connections to as many terminals as the operator requests up to the limit of the class of service.

The tactical telephone system needs to provide a means to apply and manage voice over the network while, at the same time, providing ongoing functionality over the legacy LAN. The core of the new network based tactical system is the existing shipboard-based LAN. It replaces the existing AN/STC-2(V) tactical switch infrastructure and the existing adjunct administrative telephone system. The LAN based system is a computer controlled distributive software based architecture. The proposed system needs to be able to satisfy tactical operation requirements as well as administrative functions (Figure 4). The system will have auto attendant, call forwarding, net and conference calling, voice mail, computer telephony features.

The proposed tactical system is entirely encompassed within a shipboard LAN system. The system uses the multimedia capability of the network to deliver voice to the shipboard sailor using the existing data infrastructure. In the basic configuration the POTS analog 2500 network terminals are connect to a telephony enabled interface which is then multiplexed and sent over a connection to the telephony server via the command interface. During the call setup procedure the telephony server connects the requesting and requested network terminals by way of the bi-direction audio input and output. The telephony server continues to communicate with the POTS interface using the command channeling.

The implementation of the system needs ensure quality of service (QoS) along with data on a common set of wires to each user. Voice and data are routed over the same wiring through any number of industry standard switches. Voice is delivered over specifically reserved, dynamically allocated virtual circuits within the LAN fabric. Telephony is enabled for each user via voice enabled network interface firmware. This unique architecture enables the system to have a mixture of networks and telephony services over the existing infrastructure. This architecture permits a scaleable solution and allows the ship to gradually convert to the high speed network architecture while still using the existing PBX system.

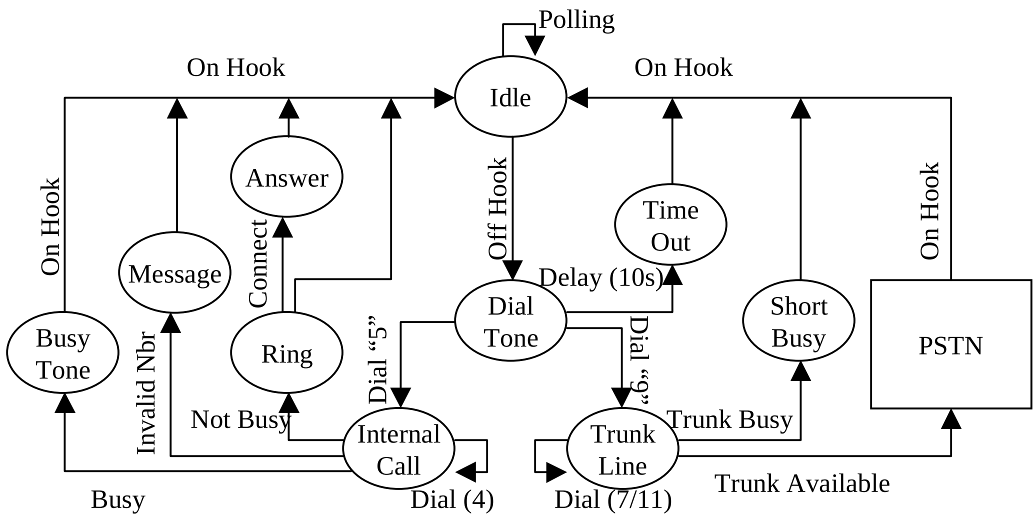

Figure 4. State Diagram For Phone Line

1. Threat Description

There are several unique shipboard threats that need to be taken into consider when developing any shipboard electronics system. Systems residing on a LAN that connect external to the ship pose additional caveats. All these conditions have to be accounted for they have to be tested against government approved standards before they can be considered candidates for implementation.

Electronic systems in use today minimize power consumption and provide a minimal amount of radio wave interference protection. It requires only a minimal amount of external electronic interference to disrupt the system’s internal signaling. The hull of the ship provides a measure of protection against external electronic interference from identified hostile forces. A significant number of disruptive forces emanate from causes internal to the ship. These disruptive forces can be benign in their origin or they can be an inadvertent attempt by friendly forces that causes signal failure. All but a few of these threats can be safeguarded against to a relatively high degree of confidence using low technical safeguards. The greatest threat, and the most difficult to guard against, is the friendly force member who is predetermined to violate the built in precautions. The tradeoff becomes: how much liberty over system and configuration control should be built into the system and how to modify and retain configuration control over those aspects the ship's force does not retain control over.

Because the system is designed to reside on the ship's network backbone and coexist with other COTS systems additional security risks are apparent. Denial of services, whether deliberate or inadvertent from internal or external sources, form the greatest possibility of system failure. This type of attack is relatively simple to accomplish and can be mastered from external sources with very little information regarding the internal operations of the system.

Improper system configuration can cause excessive chatter on the network causing, in effect, the same denial of services as if it were an external attack on the system. Prevention of this type of denial of services can only be limited by a highly trained and competent work force that routinely checks the system for identifiable problems and is knowledgeable in how to quickly isolate the offending equipment and reconfigure the system on the fly during critical situations.

Future requirements for the tactical voice communications system is the combining of secure and non-secure communications over the same voice communications system. This presents additional security requirements that need to be considered when designing the system. Methodologies employed in digital communications allow signals to couple on top of another that does not diminish communications capabilities or signal quality. This allows inadvertent coupling of signals that are in relatively close proximity to each other. Within the confines of the ship this coupling has minimal effect on security. If the communications is transported off the ship by any method, there is a significant possibility the non-friendly sources can receive the signal and decipher not only the original but also, the coupled signal. Care needs to be taken to ensure that non-encrypted classified signals do not come in close proximity to any system that has not been tempest approved.

2. Interoperability

Interoperability is the ability to utilize network terminals (Table 6) with differing features and functionality from varying manufactures and has them plug and play throughout the communications system employing a common interface and protocol standard. Interoperability between current, next generation, and commercial phone systems only exist within a limited scope of communication environment. The functionality of the network terminal is interdependent on the operations of the PBX switch. This makes most network terminals on any system proprietary to that system. Interoperability becomes particularly important as the number of service personnel aboard the ship is reduced and the technology training of those personnel is reduced to a minimal acceptable level.

Points of clearly defined demarcation are the primary methodology of successfully integrating competing manufacture's equipment. Eliminating the interdependence of the network terminal to the transport medium would create a system where functionality is designed network terminal to network terminal. The functionality of the inter-linking network would encapsulate the communication signal and primary transport medium and convey the data across the network. The receiving network interface would then perform the reverse process, creating a virtual point to point communication.

Developing the voice over LAN system using the ISDN as the primary interface would allow the inclusion of the major terminal types with only software as the principal difference. The stacks, protocols, and physical layer of the interfaces would remain identical across the different network terminals. The ideal development would use identical firmware for each of the network terminal interface with either a manual or software configurable determination.

Terminal Type	Usage	Interface Type
ISDN		
Commercial	Administrative	ISDN 2B + D
Secure Terminal Equipment	Administrative	ISDN 2B + D
Harsh Environment	Partial Tactical	1B ISDN
Below Decks	Full Tactical	2B + D ISDN
ANALOG		
POTS 2500 Handset	Administrative	Analog Low Impedance
Jack Box	Tactical	Analog Low Impedance
Sound Powered	Tactical	Analog Low Impedance
Announcing System	Tactical	Analog Low Impedance
Radio	Tactical	Analog Low Impedance

Table 6. User Network Terminals

3. Scalability

Established PBXs switches have limitations in their scalability. Expanding beyond the hardwired limits of the switch usually requires procurement and integration of additional switches at a significant cost increase. As telephony requirements increase, the system should not require massive replace upgrades as are common with traditional PBXs. The network components should provide for increased requirements without a major impact on the existing system.

Any telephony component establishing communications on the same telephone system network should be designed so that it is aware and uses the same configuration system. Although only one component may be involved in establishing the call, several components may exists and operate on a load sharing relationship. This will also allow for system redundancy and granularity. This will allow resources to be added at any location in the network. System requirements can be located at the most appropriate place in the network. The mixed scalable architecture needs to permit organizational flexibility to allow for deploying various types of telephony systems while still retaining existing LAN components.

4. Granularity

The load sharing and co-processing abilities, the proposed development should be developed and implemented in a granular methodology. The development of features, wherever possible, should not be developed requiring co-dependency on existing features. The telephony network components need to be distributed across the ship on as required by each location. This requires each controlling aspect of the system to be aware of all other peer level controllers. Rather than a hot standby type of implementation, each controller or telephony server, is required to shoulder an ongoing portion of the load. As the system increases in size, additional servers can be placed online, configured and automatically assume the proper balance of the services.

5. Redundancy

The shipboard tactical telephone system requires that the system be available for use at all times. The existing tactical switch has benefited from decades of use and support. Reliability and redundancy should be built into the system. The system does not require centralization and should be designed to eliminate any single point of failure. The network supporting the development needs to be inherently fault tolerant, providing a robust telecommunications system. Any system replacing the current tactical phone system needs to meet or exceed the current reliability standards. The system should be designed to avoid architectural bottlenecks.

6. Standards-Based Telephony

Except for a limited number of network terminals and certain features of interface interoperability, most PBX switches do not maintain any interoperability between commercial systems. Some of this has to do with operational, functional or other technical features, but most of it is economic based reasoning. When a system is standards based then manufacturing of components becomes a lowest bidder effort and it becomes difficult for the manufacture to recover development costs and to make a significant profit. These are the same reasons why the Navy should base its development of industry standards and only incorporate COTS equipment that does follow standards.

The incorporation of standards base in the development of the system allows for the integration of multi-vendor equipment across the switch. This allows the Navy to focus the development on equipment performance specifications and not on the internal detailed operations of any specific equipment. This type of development is based on the assumption that there are several developers that manufacture any specific piece of technology. Because the Navy has unique requirements and a limited market, there are insufficient resources within industry to totally support COTS or commercial standards based development. The most effective approach is a hybrid system that employs industry standards and COTS, when the equipment is commercial available, and internally developed components that are phased out as commercially available replacements become available.

B. SYSTEM PERFORMANCE CAPABILITIES

There are network voice communication devices in existence today that perform well enough to be given consideration for possible inclusion as administrative voice systems. However, many additional considerations need to be taken into account when making determinations for tactical communications. End to end performance capability is what distinguishes the abilities of a tactical audio communications system from an administrative system or the hobbyist's ability to communicate over the internet.

In functionality, they all perform relatively the same; they all pass audio frequencies from one point to another. It is the time and effort required accomplishing the task and the quality of service of the transmitted voice that determines what type of system has acceptable levels of reliability and performance. Even though the system is used and does have strictly administrative features, its primary function is during highly critical shipboard situations. By ascertaining the liability if the desired objectives are not obtained makes the level of determination. Since safety and the value of human life are paramount, the ability of the system has to meet a significantly higher level of performance standards available within the industry.

1. Traffic

In the existing telephone system, traffic is limited by the ability of the call control processor to service each terminal. Even when a terminal is not connected to the PBX, the CCP reserves a time slot within the computing cycle to service any, non-existing, request. Because the current technology is time sliced rather than interrupted driven, the ability of the CCP to service incoming requests is severely limited. This limitation is marginally circumvented by physically load balancing the forward and aft PBX switch processors so that neither is overloaded during critical events.

The shipboard LAN is a shared resource and there is limited ability to precisely determine the exact load of any aspect of the system at any given time period. In the new development the voice over LAN system has to have the ability to perform non-disruptive transfers of network terminals load and functions located on the LAN within specified time allocations. This may include design criteria based on load balancing due to a terminal's physical placement in a high or low utilization area. This determination is made during busy hour when the system has to service requests from a maximum number of terminals. Because it is difficult to determine the exact amount of effort required by the controlling functions at any give time, more comprehensive traffic requirements for determining load sharing and parallel processing efforts needs to be implemented. The system should be designed such that all possible combinations of requests for all traffic requests and services from all terminals located within the LAN have to be serviced by the system within the allocated bandwidth and time limit. If the system is unable to operate a minimum level of performance, the network needs to have the ability to determine priority of all the communications within its domain and reduce or eliminate communications whose priority fall below the maximum bandwidth utilization ability.

2. Blocking Criterion

Because of the tactical aspect of the switch, no service request from any tactical network terminal can be blocked at any time as a result of the switch being too busy to process the call or because of unavailable resources within or between switches. Although this is a correct statement, the implementation of the system has shown this to be impractical. The physical layout of the terminals aboard the ship limits the number of inter-switch connection requirement during any tactical requirement. If the switch gets overloaded with incoming calls the system merely queues any incoming requests for service until processing time becomes available to service that request. The current tactical system can only process sixteen concurrent requests. During busy hour this can mean that, from the time the last digit is pressed until forward ring, it can take up to four minutes to complete a service request. Although this is not a likely occurrence, the fact that it is an acceptable specification shows the system's blocking limitations.

LAN based telephony has a similar problem that needs to be overcome if it is to be a viable contender in the communications market. LAN based data communications is best effort driven technology. If the communications channel becomes too congested with communications, it just buffers data transfers until the channel becomes more available. While delays in traffic flow have minimal effect in data communications, in audio communications it can have a major impact.

The requirements for the tactical system requires there be at least two switch servers providing terminal services as well as network call setup and take down. The system is designed where each switch has an additional hot backup. This creates a system that has four full switches where only two are active at any one time. Rather than have a hot standby system, as is currently implemented, the telephony servers should have four or more active load sharing call processing switches. If one of the switches fails the other should have the ability to handle the entire load. The design should also permit granularity of the system by allowing additional switches to be added at strategic parts of the system thus further reducing average load requirements while allowing critical network segments to work in total isolation if required.

3. Call Setup Time / Setting Up a Connection

Call setup is the ability of the telephony server to receive data from the requesting network terminal and setup a connecting circuit across the LAN to the receiving network terminal (Table 7). Call setup is a telephony server function that accepts digits from the requesters touch tone pad and translates them into a network address. The telephony server has to validate the rights of the requesting and receiving network terminals as well as negotiate the privileges they have to utilized resources on the network.

The response of the telephony server to requests from the network terminals does not have a real time response criterion, but does have to perform a maximum number of requests within a specified period of time. Most of the time request for services from the telephony server will be minimal. There are peak periods of time when an overwhelming number of users will attempt to request services. During this peak period call processing and setup of the services has to appear to the user to be in real time. The telephony server should be able to handle up to thirty concurrent network terminal requests without any noticeable delay in response. Because it is expected that a first-in-last-out interrupt driven stack will be required to service requests, as the number of concurrent requests for services exceeds the available real time processing cycle, a noticeable delay of a few seconds will occur. If the delay becomes excessive, elements of the granularly of the system design should be implemented.

4. Automatic Route Selection

Automatic route selection is the ability of the network controllers to determine the shortest time path for the voice data to transit across the network. The network controllers continuously monitor the network for bottlenecks, circuit failures, or other signal degradation causes and has the ability to dynamically reroute traffic. The LAN based telephony system has to have the ability to vary the actual route of any aspect of the communication and can vary with each instance of the call. This allows the system to compensate for network changes, circuit availability, and system failures.

The LAN based system needs to maintain, at least, a single layer of redundancy once the communications is established. If for any reason the communications channel should fail the system should provide hot cut over capabilities. The cut over of the NT should be transparent to both the NT user and system administrator. There are several types of networks available in the commercial market that provide hot cut over, or the equivalent, services as an integral part the network environment. Unfortunately, the most popular network employed in the Navy does not have automatic fault detection and correction.

The initial call signaling command to set up the dedicated circuit within the LAN is sent to the LAN controller from the telephone server when the receiving terminal goes off-hook. Once the call is established between the network terminals the LAN controller is responsible for maintaining and making any circuit changes to the connection until all terminals go on-hook.

5. Warm Reset Time

The warm reset is defined as the time it takes for the system to reach fully operational condition from the time power is first applied to all components of the system. This is how long it takes the system after a complete system shutdown to reboot and reestablish all circuit connections to the last know good condition.

The warm reset time is based on the assumptions: that the ship is in battle emergency or otherwise critical situation; that the tactical voice communication system has lost part or all operational ability do to events external to the system; that re-establishing communications is paramount to the safety of the ship's force, that any delay in voice communication could result in the loss of life or property; and that the ship's operational capacity in a critical situation. This presupposes that all system components are operational and that all other tactical environmental conditions have been met.

The reactivation of the system requires the telephony server(s) to maintain an active database of all connections as they are setup and taken down. When the system goes off line if there is an appropriate shut down procedure that is annotated within the system database. When the system is re-activated it checks the status of the last known condition of the system. If the system terminated in a non-standard state the re-boot procedure assumes that a threat conditional exists and reprioritized boot procedures giving preference to critical communications circuits over standard maintenance or house keeping procedures.

The reboot process reestablished all network terminal connections to their last known good condition. Circuits will not be reestablished for those network terminals that have gone to the on-hook status subsequent to system shut down. There is a window of lost opportunity between the time a network terminal goes off-hook and before the circuit is established. If the system shuts down during this time period, the terminal connection information is deleted and the call is treated as a terminated attempt. The attempt has to be re-initiated from its inception by going on-hook and re-establishing the call.

Traffic	Description	Criteria
Call Busy	The number of telephones unable to make or receive a call because of network bandwidth limitations	No more than 0%
Call Initiation	The delay from going off-hook to the receipt of the dial tone.	3.0 seconds
Call Completion	Delay from the last digit dialed to ring of the receiving phone.	0.5 seconds
Call Blocking	The number of phones unable to completed a call because of a control subsystem error, false switching or erroneous station signal.	0.001
Call Misrouted	The mis-routing of a call due to hardware or software error.	10^6
System Failure	When 15% or more of the phones do not have the ability to originate or receive calls.	0.999995

Table 7. Busiest Hour Traffic Handling

IV. INTEGRATING TACTICAL VOICE OVER THE SHIPBOARD NETWORK

A. OVERVIEW

Deployment of voice over data networks is at the heart of the convergence requirement. The new system must not only provide an effective alternative to circuit switched telephone systems, the consolidation must also save shipboard time and resources. The implementation should be seamless and must incorporate the traditional PBX functionality across a ship-wide local area network. To the end user, the operation and functionality of the LAN based tactical telephone system must be identical to the current tactical system.

The system must incorporate all current external trunk line access and provide service to the tactical phones, emerging technology and traditional POTS analog network terminals. The new system should incorporate functionality not currently incorporated in the tactical PBX such as system administration and management, voice mail, call forwarding, call waiting, and other feature commonly available in commercially PBXs.

The tactical telephone is required to co-exist on the network infrastructure with other tactical and non-tactical systems. Because of the nature of the infrastructure this poses significant security risks not normally associated with stand-alone systems. To overcome this limitation the communication system needs to provide a trusted relationship between the communication terminals and the public accessed network.

B. BROKERING

In traditional telephone systems the telephone switch acts as a central processor unit (CPU) through which all communications control commands are processed. The processing time of the CPU is evenly divided between each of the logically connected terminals. Each terminal is allocated a maximum amount of processing time and no single terminal can surpass its time allocation. Any time slice not fully utilized by the requirements of the terminal is cannot be reallocated back for system administration, "house keeping" and other non-time critical functions. This method of communications provides the system with direct control over the communications channel of each and every terminal on the system and controls all resources within the telephone network.

In voice communication over a shared networks, each terminal has to have the ability to directly communicate with other network terminals located within the network and to external PBX switches through the trunk-line interfaces. The system has to have the ability directly address the location of a given terminal based on a phone number or the terminal has the address of a centralized server that provides this database services (Figure 5). This trinary communications interface requirement is located within the local domain of the shipboard network and has to provide a proficient method for metered services. Because the system will reside on a network, where it has to compete for resources and services with users other than the tactical telephony system, a method of resource brokering needs to be established to allocated public assets, determine interface protocol compatibility, enforce class markings within the terminals, prevent unauthorized terminals from gaining access, and prevent denial of services attacks. The combination of the trinary interface and the telephony server, through the network infrastructure, provide the fundamental control and communication features of the telephony based tactical voice communications system.

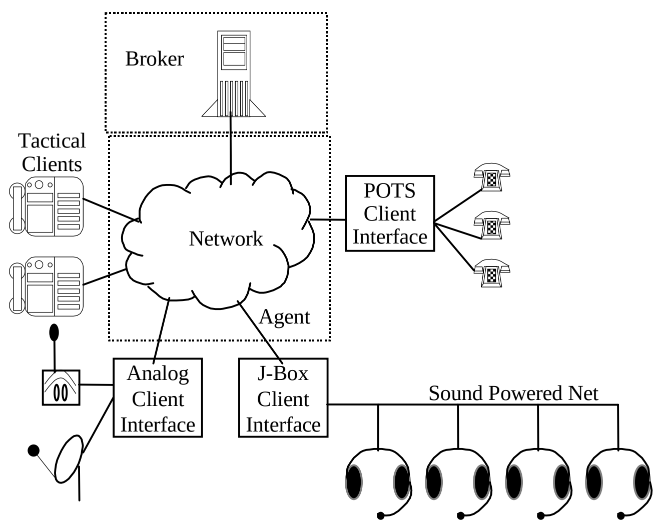

Figure 5. Brokering Management

C. RESOURCE CONTROL AND ALLOCATION

The tactical voice communications system will be deployed in an environment where the resources are shared and available to users of unknown origin. The system has to have means of controlling owned assets while preventing non-affiliated co-residing users from gaining access or preventing legitimate users from gaining access to assets. The system also needs to have a system of brokering shared resources in such a fashion that minimum performance standards are met for both the tactical communication system and other critical users operating within the environment. The problem is further compounded because the owned resources are physically separated from each other and logically connected by the shared, non-owned, resources.

The optimum implementation would be using a broker between tactical telephony resources and employing agent for gaining control common resources. When a tactical network terminal is idle it competes for network resources at the same level as any other user.

One of the first resources is a dial tone (Figure 6). It is the function of the telephone server to ensure the request for services is coming from a legitimate source. This is not a highly time critical requirement, but does have a finish within limitation of a few hundred milliseconds. For the terminal operator this is the dial tone response, for the system this is when the terminal requests permission to access resources within the tactical telephony system. This exchange sequence identifies the network terminal to the broker who then verifies it against an internal database as a legitimate resource. At this time the broker authorizes the terminal to request use of tactical resources. This is not an authorization to use tactical communications resources, but an authorization to request the use of NT resources. Using a broker to identify the tactical resource prior to usage permits the system to provide administrative functions and prevents non-tactical users from accessing the system or effecting denial of services.

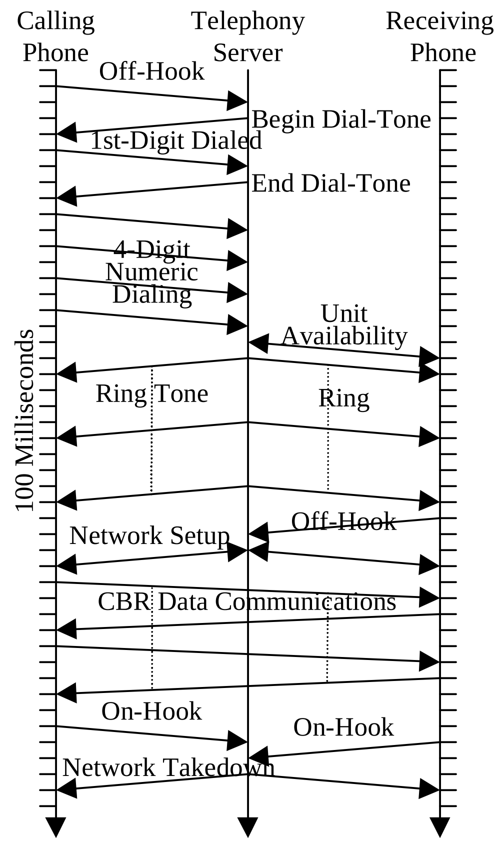

Figure 6. Event Trace For Phone Call

Once the broker has verified the identity and configuration of a terminal, that terminal is then authorized to request the use of other tactical resources. When requesting a dial tone the terminal then dials a phone number which is sent to the broker which checks the number against a database for verification that the requested number is an owned resource. If the requested number is an owned resource of the broker, the broker sends a message to the terminal requesting services. When the terminal receives the request it responds to the broker with a warrant for a request services. Although it shortens the procedure if the broker were to send out the information to the receiving terminal, this would compromise the security of the system. The owned resource always sends the warrant to the broker, the broker checks the information against an internal database then grants or denies the request.

Up until this point all requests, messages, and warrants between the network terminals and the broker are exchanged across the data network with only slightly elevated priority compared to any other user competing for bandwidth allocation across the network. Once the broker determines that both the requesting and requested terminals are legitimately owned resources and are available for communications the broker then requests a high priority constant bit rate allocation be set up within the data network backbone. The data network is not owned by the telephony system and there are many users of differing priorities requesting services on the network, therefore all requests are made through an agent of the broker to an agent of the network.

The request for services is negotiated with the network with sufficient information from the telephony broker to the data network to determine where the request for services falls within the existing users on the network (Figure 7). The data network has several possible options that have to be negotiated to determine if there is available bandwidth to set up the channel as requested. If the network does not have sufficient bandwidth for the requested allocation but does have users with lower priority than the broker's request the network agent has to determine which users are de-allocated bandwidth and how to accomplish the task. If a request made by the telephony server for additional bandwidth and the network does not have sufficient bandwidth available, the system has to have sufficient priority to bump existing channels on the network or it has to return an unavailability token to the broker's agent and let the broker determine what to do next.

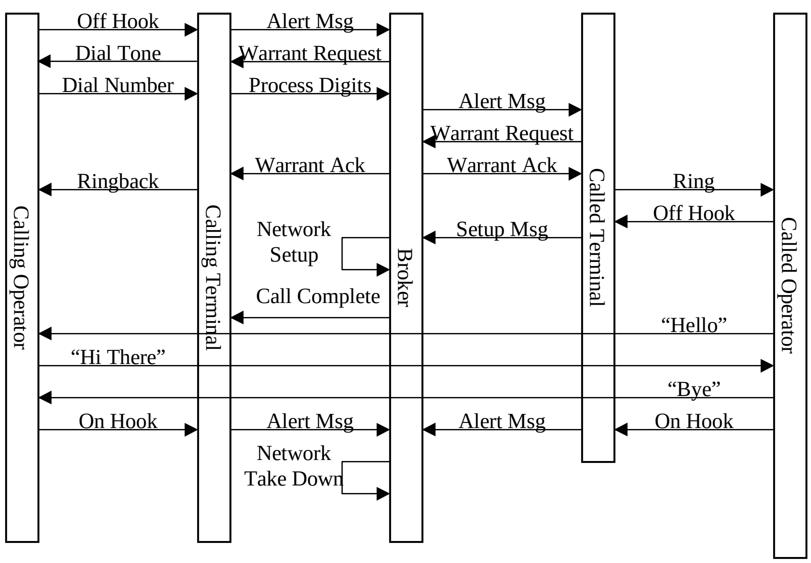

Figure 7. Call Setup Interaction Diagram

D. COMPUTER TELEPHONY

The current tactical switch functions by dividing the processor time into time slices. During each time the processor takes the digitized signal from the incoming voice communication and places it in the location of the outgoing signal. The processor then sends out to the terminal any signal that may be in the out processing queue. This method of signal processing limits the system's total number of telephones the CPU has ability to handle, but provides a tightly controlled series of signal handling processes.

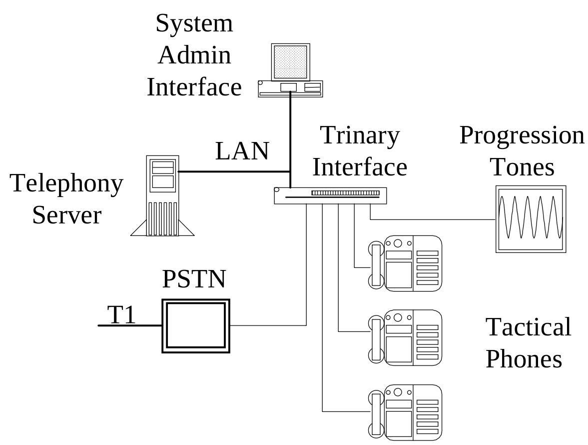

Figure 8. Basic Tactical Telephony System

Quality of service is the most significant problem in dealing with LAN telephony. The system changes from a tightly controlled centralized PBX switch to loosely coupled procedures and processes where priorities are not conjoining (Figure 8). In a data network there is limited synchronization, or ability to synchronize distributed processes. For the convergence of voice and data communications to be successful the end points of the voice communications have to have the ability to meter QoS and to signal the broker if the standard is not being met. The broker, or telephony server, then has to have the ability to control services to meet the priorities of the overall communications requirements. In addition, every process at any point along the communications course, however fractured it may be, has to agree to accommodate the priorities.

By implementing a voice over LAN approach, and specifically a tactical voice over ATM, many of the design challenges are resolved within the design specifications of the network. The telephony server institutes a connection between or a disconnection from external devices as a result of an operator or radio action. The telephony server performs this action based on configuration data stored with in the system and modified via the system administrative interface. The telephony server maintains the network's current status and reflects the current status of every device within the LAN tactical voice environment.

Combination of the telephony server (broker), trinary interface and progression tone server provides the cornerstone of the traditional services of the PBX telephone switch within the LAN based system (Figure 9). These three elements added to the data network topography are required to provide all the basic tactical switch PBX features for the merger of voice and data that are provided on the traditional phone systems. The new combined broadband system has to appear in usage and functionality to the user as a traditional phone or data system. The fundamental components can be added to an existing network to create a relatively inexpensive yet powerful system. Using existing analog and ISDN phone interfaces as one of the trinary interfaces, the telephony server can be used as a stand alone phone system with the use of a single broadband network switch.

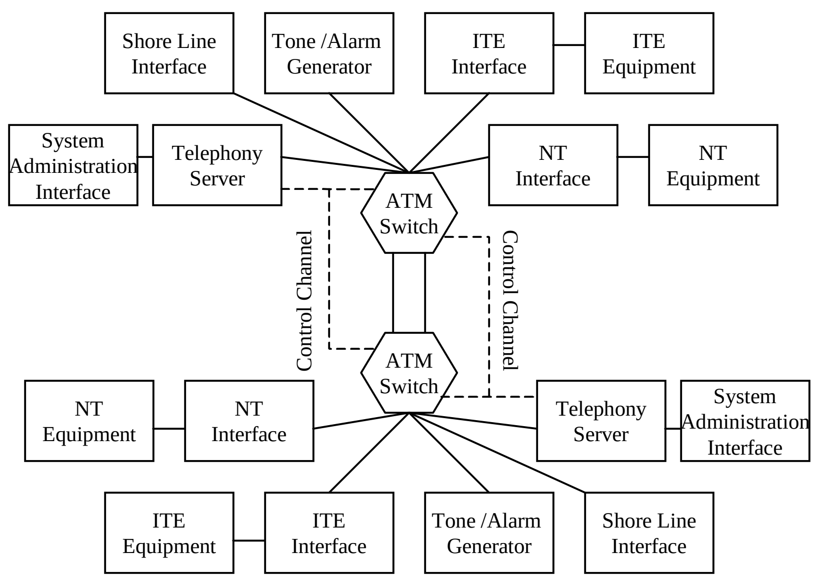

Figure 9. Tactical Telephony Block Diagram

1. Telephony Server

The telephony server along with the trinary interface and progression tone generator provides the services traditionally supplied by the PBX. These three devices are tightly coupled in their operations, but are required to interface over a loosely coupled LAN. Most of the services provided by the telephony server are non-time critical. There are specifications for the server to prove certain services within a maximum allocated time, but variances of up to a few seconds are acceptable.

Unlike the traditional telephone system, where there is a single server allocating all resources, a combined system will have several classes of non-related users sharing a common infrastructure. Because it is not a closed system, it is assumed that most of the common resource users will be unidentifiable to the telephony portion of the system (Figure 10). Before any user is allowed to gain access to the system the telephony server has to verify authenticity of the network terminal. The authentication is required to prevent users of the network of lower access rating to tax system resources beyond their allocation. The telephony server then brokers the network terminal's request for services over the broadband and local area networks.

In addition to authentication and classification, all users of the system have to be prioritized to allow an overarching LAN manager to deny, delay or reroute service requests of all system users. The telephony server works in conjunction with the management control features of the network resource manager interact in a broker client relationship to provide these services. This allows the telephony server to provide control features that monitors the quality of service for each of the active network terminals and sends requests to the network for reallocation of resources when QoS is not being met.

One of the primary functions of the telephone server is to provide dialing services to the network terminal. The dialing services converts the instructions given through the network terminal's human interface panel into commands understood by electronics of the system. The converse is also true: dialing service converts the instructions and commands related to the electronics of the system into signals understood by the human operator. Dialing services requires the inter-linking of the progression tone generator to the user's terminal across the LAN. It is the progression tone generator that provides the human interface functionality conveying the status of the telephone system.

The most critical aspect of the telephony server is the ability to monitor and control the network terminal's quality of service. The ability to monitor and control QoS for the terminal requires the trinary interface to observe signal quality as it is received. Because the trinary interface has no ability in itself to exercise control over the network, it has to notify the telephony sever of the conditions of the incoming signal. The telephony server then has to negotiate or broker with the network manager for reallocation of the network resources. The telephony server has to have the ability to interface with all network terminals in a near real time basis when brokering the bandwidth allocation between the network terminals.

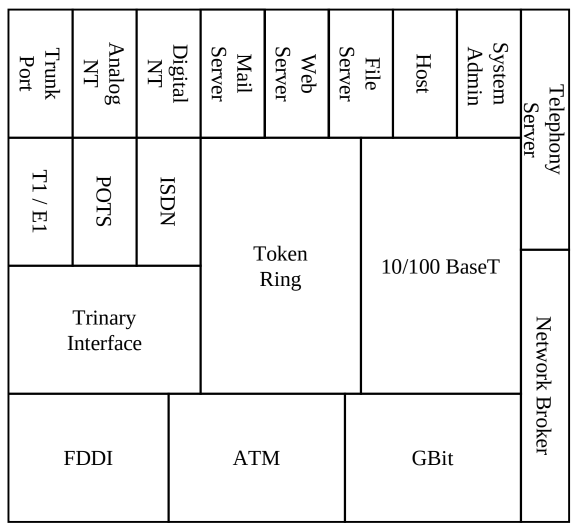

Figure 10. Voice And Data Convergence Over Broadband Block Diagram

2. Trinary Interface

The trinary interface (Figure 11) is the key feature that allows the network terminals with none-LAN interface standards to communicate with other user terminals utilizing LAN resources. The trinary interface along with the telephony server is what provides the services over the LAN based telephony system that is normally provided in the traditional telephone system. When PBX telephone goes off-hook the switch sends out a dial tone to the terminal. There is no equivalent in the LAN based systems for providing progression tones and other services normally expected in phone systems. The trinary interface is the means by which the system can provide these services.

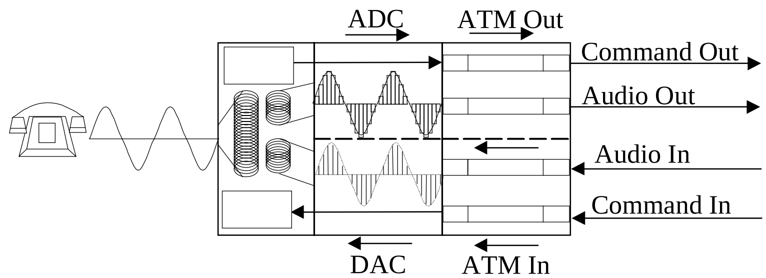

Figure 11. POTS Trinary Network Interface

Interfaces to the LAN receives signaling in data format, usually asynchronous non-real time form, and have to re-order packets. The packets are then stripped of header information and the remaining data is sent to the network terminal interface. The terminal interface buffers the data, adds frame timing and repackaging the data depending on the terminal type. The synchronous packet is then sent out to the network terminal. The trinary interface has the ability to tell the telephony server that bandwidth requirements are not being met and to request the LAN to shut off non-essential services of lessor priority. This give the trinary interface the capacity to determine if quality of service standards are being realized and to signal the telephone server if they are not.

One of the primary functions of the trinary interface is the ability to characterize and route commands coming from the network terminal to the proper destination. The trinary interface has to know when to send commands to the telephony server or as data out to the network interface. Many of the commands coming from the network terminals are indistinguishable in form from data to be passed across the network or commands to be passed to the telephony server. Some commands are always routed to the telephony server, other commands are only routed to the server during specified times of the communications cycle. The primary distinction is the nature of the command and at what time during the calling cycle they occur. The trinary interface has to be able to discern the difference and properly route the command.

The trinary interface will require the network terminals to negotiate an acceptable format for terminals of different communication format types (Figure 12). In conventional systems this is not a major issue because all in coming call formats are normally converted into a common format at the switch then reconverted to the outgoing channels format before being sent to the receiving network terminal. Because there is no common switching in the proposed system, the ability to negotiate a common format between network terminals is placed on the trinary interface. The classification communication between the NTs has to occur prior to establishing transmission in order for the telephony server to request from the network broker the required LAN resource allocation.

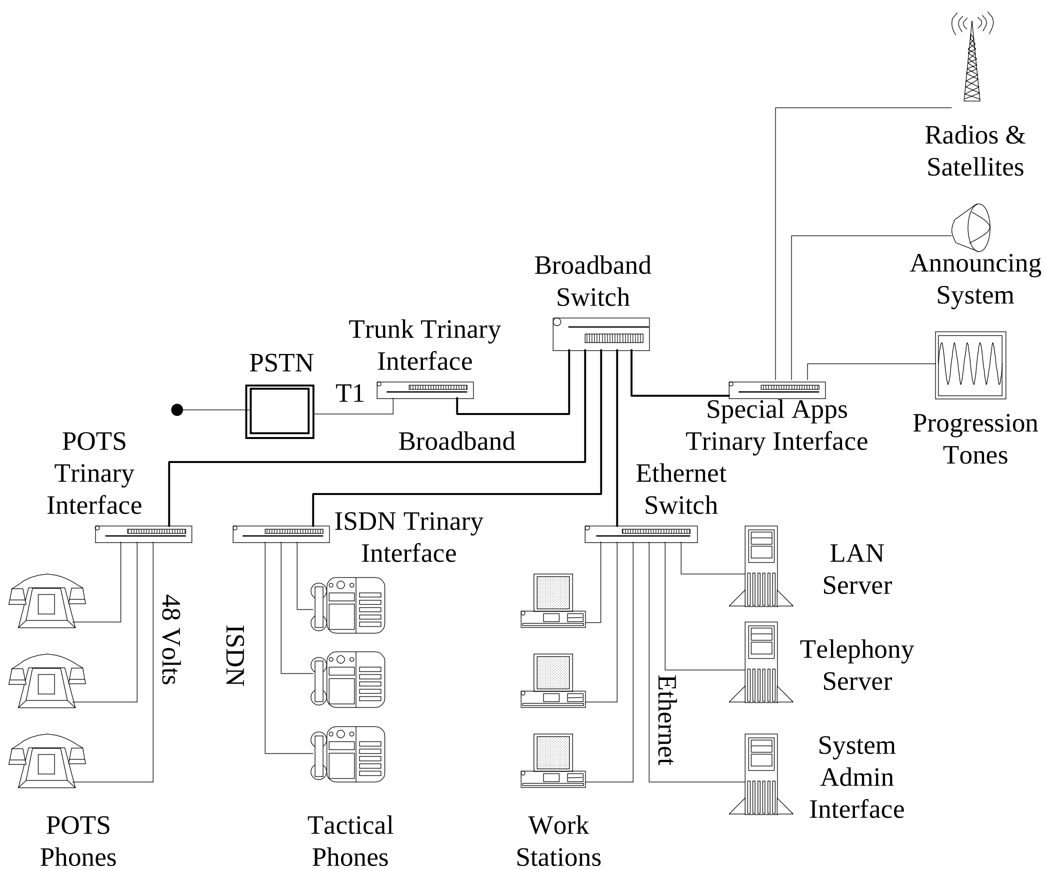

Figure 12. Network Centric Tactical Telephony System

Placement of the trinary interface is critical to maximize the benefit of a common data and voice infrastructure. By integrating at the base of the network terminal the trinary interface with the user terminal allows the system to deploy a common cable plant and will allow maximum interaction with the telephony server to ensure QoS within the interconnecting network. Due to non-software related constraints it may be required to place the trinary interface in a physical location other than next to the user terminal. This creates unique wiring specifications for the data and voice terminals, but is still a viable solution.

3. System Administration Interface

The system administrator interface (SAI) is how the system operator monitors and controls the functionality of the telephony server, network terminals, and other configurable system devices. The SAI is how the system administrator configures the telephony server and programs all network terminals. The SAI also assists the system administrator in diagnostics of the system when automated diagnostics are unable to detect and notify the system administrator of the problem. The SAI is an independently operated computer interface and is only employed for the configuration and monitoring of the system and does not impede operations of the system should the SAI become non-operational.

4. Internal Diagnostics Testing

An integral part of the telephony services is the ability to detect faults within the system and seamlessly reroute the call or switch to a hot standby backup. The system has to be able to detect system faults as soon as they occur in order to prevent loss of sign or signal quality. The system should also prevent false connections due to unpredictable events as well as continuously validate all current connections. The auditing of the connections can be preformed against a connection map that compares current connections against actual connections.

The system needs to perform on going tests of embedded software and hardware in the external terminal interface cards, backplane I/O channels, within the network terminals interfaces and other end equipment. No single point failure should cause a loss of more than ten percent of the total communication capacity. For subsystems whose failure would cause a loss of more than ten percent of the system capacity, a hot backup or rerouting configuration should be employed. When an error or failure is detected the system should seamlessly disable any elements on the previously active component that would inhibit the proper operation of the switch. A normal course of development for the backup scenario would be to have a primary control unit actively monitor the operations of the active units and when an error is detected aggressively take control and shut down the suspected defective unit and reroute any active signals.

The system should be designed in such a manner that all devices or sub-systems within the critical path of any tactical call can be hot swapped without the interruption of any active call. For most devices the hot swappable ability will require that the power and signals be applied to the card level components via the backplane or I/O interface. The component or subsystem being hot swapped should require a minimum of configuration setup by the technician or operator. The technician should be able to place a device within the system and that device should have self-configuring ability by determining its location within the system and reading configuration tables of adjoining devices.

When a fault is detected the system will notify the operator via a series of notifications or alarms based on the criticality of the failure. (A notification is simply informing the operator of a system condition that is not expected but does not affect the proper operation of the system.) Terminal equipment improperly identified within the database that does not match the signal type expected from that terminal would be an example of a notification. A minor alarm is when a sub-system fails to operate properly and the subsequent outage is less than ten percent of the total capacity of the system. A major alarm is when the failure affects greater than ten percent of the total operational capacity which would normally require a hot standby to assume operations.

5. Hot Switchover

One of the more difficult software functions to implement on the tactical switch is the hot cut over of any out of service equipment without disruption in service or loss of call connection. The primary method of determining system failure is by having the active standby perform the identical function as the active component plus additional monitoring functions. The standby unit compares the output of the active unit against its own output. If there is no comparison, the hot standby has to make a determination if the error is its own or if the failure is the active unit. If the failure is with that active unit the standby has to disable the active unit and take over as primary unit.

Because the current tactical system has to prevent single point failure logic, a double fault in the active and hot standby could possibly disable the entire switch or onehalf of the telephone system. In the new LAN based system, the telephony server does not handle the entire call control functionality. Rather than a hot standby with multiple redundancies, the voice over LAN should employ a load sharing method of backup. This means that several system components have to be aware of the status of the entire system but may only handle a limited portion of the service requests. If a unit fails then the device(s) sharing the load assume the responsibility for handling the burden of the failed device. The converse in also true: if additional devices are placed within the system they should be auto-detected and the interchangeable devices provide current configuration requirements and the system should load balance the work responsibility.

The deployment of this type of development offers several distinct advantages. The development could be granular in nature. If the system is small enough in functionality and size a single telephony server could handle the entire system. As the requirements increased, the functionality could be passed off to a parallel processor(s) along the LAN without an increase in development effort. As the physical size of the system increased, the system could be developed in a load-sharing configuration. The load sharing would allow identical units to be mounted on the LAN. When the new unit made its presence known existing system units would determine the location and ability then configure the new units for optimal operation. This could also work during a failure analysis; when the system failed one or more identical units could vote on the failure of the system and if it was determined the unit was no longer available for service could assume the services of the failed unit. This would require each unit on the system to maintain a configuration map of the entire tactical telephony system and have the ability to, in a timely manner, reallocate loads amount the remaining properly function units.

E. STANDARD NETWORK TERMINAL INTERFACE.

In the commercial communication world there are very few commonly acceptable standards that allow network terminals from one manufacture to communicate across the interface of a competitor’s system. As terminals evolved from the basic analog into the multi-vendor form, the interconnection has been extended to include a network terminal interface that matches the network terminal protocols to the PBX switch. The network terminal line interface is what connects to the network terminal into the PBX. In the ideal situation there would be a straight connection from the network terminal into the PBX switch.

Developing the LAN based system will require each network terminal interface to function as its own switch and employ the telephony server to provide brokering services between the terminals. The normal development would be to provide the same services to the network terminal that is provided by the native network services. Implementing a development utilizing this format would preclude the use of any third party terminals or interfaces.

To take advantage of third party COTS and unique developments, strategy needs to be implemented that utilize the optimum feature of both technologies as well as allow the functionality of the COTS network terminal to communicate directly with other terminals and interface with, and obtain the services of, the telephony server. The trinary interface can talk to the network terminal in native format while encapsulating the data across the network in the networks primary format and providing a third interface to the telephony server for overall control and brokering of the required services.

F. NETWORK TERMINALS

The network terminal (NT) provides the means by which the user requests services and communicates over the system. Because the subscriber equipment is not LAN compatible it has to be linked by the subscriber interface. The network terminal interface provides this connection for the individual telephone unit into the LAN backbone. It connects to one leg of the trinary interface, which converts the class of service of the NT into the protocols required by the telephony system.

There are several classes of subscriber or dial up services that are connected to the tactical switch through adapters. It is possible, but not recommended, for the initial design that the terminal interface unit be developed with single-protocol terminal interface capability. Although this would provide a cost saving in maintenance and installation the saving would not be offset by the increase in design and development when additional protocols are required.

1. Plain Old Telephone (POTS) Network Terminal

The analog 2500 handset network terminal, commonly referred to as plain old telephone, is the primary non-tactical means of communication. Although this type of telephone has been in existence for many years, it is no longer compatible with most modern communications systems. However, the implementation of the tactical telephone system needs to be able to handle voice traffic over analog 2500 handsets. To the user this telephone appears to be identical to any COTS terminal within the terminal class. Beyond the initial functionality of the analog signal of the phone, the signal needs to be converted into digital format and packaged within the protocols of the LAN. In a normal POTS system this functionality would occur with the confines of the PBX. On the voice over LAN system the conversions have to occur prior to integration on the network. Because of the relatively simplistic nature of the analog network terminal several interfaces can be multiplexed into a single network terminal interface thereby conserving real estate within the network switch.

2. Sound Power Phone Network Terminal

Sound powered telephones are the ship's primary communication method during tactical shipboard situations (Figure 13). Any terminal on the tactical voice system, with the proper permits, can communicate on the sound powered net as if the net was a terminal on the system. The sound power net has an addressable phone number as if it is any other network terminal, but call answering is automatic without a forward ring or ring back. An operator on the sound powered net does not have the ability to dial out of the net. The sound powered telephones can not initiate calls. Once a network terminal links into the net it would appear to all other to be an integral part of that net with all the rights and privileges.

At no time can the failure of proper operation of the technical voice communications interfere with the operation of the sound powered telephone. If the entire tactical voice system fails the nets created by the sound powered telephone system must remain operational, or at least, the failure of the tactical voice system can not cause the failure, in whole or part, of the sound powered net.

The physical connection to the sound powered net is considered a special connection terminal. The sound powered system is always considered off-hook and any terminal, with the proper class marking, can enter the circuit. Because of the high background noise, any terminal allowed to enter the circuit must be equipped with push-to-talk features. The signal conversion and matching is accomplished beyond the normal confines of the sound powered system, but prior to being placed on the LAN.

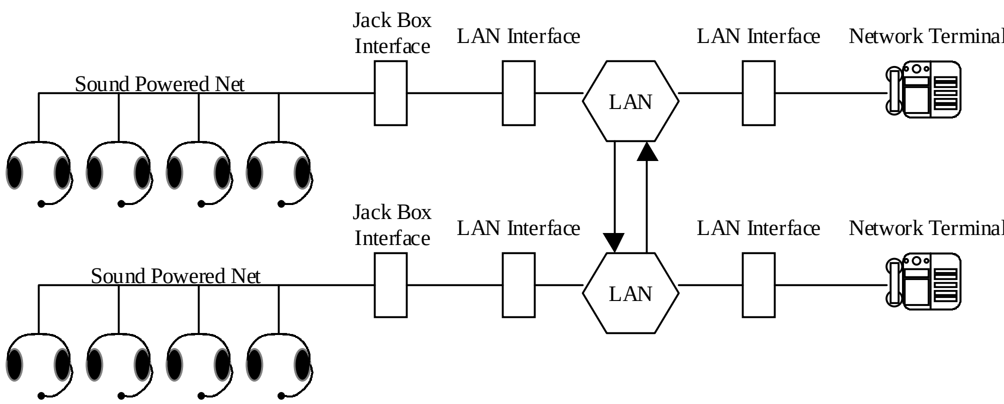

Figure 13. Sound Powered Connection Diagram

3. Primary Tactical Terminal Equipment

The primary tactical terminal is the only fully functional tactical communications network terminal. The primary tactical has the highest level class of service available to any network terminal. The class markings are assigned to the terminal by the system administrator and are the only limit the NT's operational capacity (Table 8). The primary network terminal is a unique communications device that, unlike other network terminals, requires collaboration with the PBX for proper operation.

The primary tactical network terminal has the only true non-blocking network terminal. It has to be able to accept all incoming calls from any terminal with the proper class markings. The disposition of any incoming call is at the discretion of the terminal user. The incoming calls can be placed in one of four status situations: it can be placed in a stand alone communications, placed in standing tactical network, placed in a meet me network, or placed in an existing conference net. If the incoming call is already connected to a network then all users on both networks are fully interconnected.

The primary tactical terminal also has the ability to select and de-select any channel for active incoming or out going communications. The system operator may determine that certain communications requires passive listen mode only while only select channels require active outgoing communications ability. These options are selectable channel by channel and switchable at any time during the communications cycle. Coexisting with the selectable communications provides the ability for the operator to activate out going communications at selected times. This ability allows the user the ability to passive listen to incoming communications without injecting any background noise or interference unless there is a desire to communicate.

In the traditional PBX the signals are summed within the PBX itself and the tactical network terminal serves simply as a mechanism to switch between summed channels. Because the tactical voice over LAN will not employ a device that can service a comparable PBX function. The network terminal or trinary interface will have to supply some of the functionality currently provided by the PBX. This approach dramatically alters the fundamental allocation of network bandwidth. Where previously the terminal had a single bi-directional channel into and out of the PBX and the maximum utilization could be easily calculated and accommodated for, the new process of communications means that it is possible for the tactical terminal to setup a communications channel with every communications device on the network. This will also require that summing of the conferencing calls will take place within each individual network terminal connected to the net or conference call.

Call Operation	Terminal Type						
	ISDN Tactical	ISDN Harsh Environmen	ISDN Commercial	ISDN STE	POTS	SPT	Jack Box
Call Transfer	X	X	X	X	X		
Abbreviated Addressing	X	X	X	X	X		
Tactical Net	X	X	X	X	X		
Meet Me Net	X	X	X	X	X		
Progressive Net	X	X	X	X	X		
Privacy Override	X	X	X	X	X		
Call Forwarding	X	X	X	X	X		
Call Waiting	X	X	X	X	X		
Secure Comms	X						
Call Monitor	X						
PTT	X	X				X	X
Auto Answer	X					X	X
PA Connections	X	X					
Shore Line	X	X	X	X	X		

Table 8. Network Terminals Operation Features

4. Secure Terminal Equipment (STE) Network Terminal

The secure terminal equipment (STE) is the direct replacement for the secure terminal unit (STU). The STU uses only the analog POTS interface into the system. The STE can use either the POTS or ISDN interfaces into the telephone system. The ISDN is designated as the primary interface to the STE and is required to utilize the full functional capacity. Both of these network terminals are commercial off the shelf components and have no ability to be modified to meet alternate telephony configuration standards. To interface properly within the tactical voice over LAN environment, an interface needs to be developed that emulates the required STE signaling and protocols. The interface needs to have the ability to connect and convert the STE protocols of the ISDN into the LAN formats as well as all the control and monitoring functions of the traditional PBX.

G. TRUNK LINE INTERFACES

The telephone network backbone requirements for shipboard use have not changed significantly over time. Time division multiplexing (TDM) trunks are still the primary method of transporting multiple circuits over a common carrier. However, advances in LAN technology have progressed to a level that TDM trunks can be replaced by virtual trunks carried by packet switched networks.

Use of various voice encoding algorithms make more efficient use of the available bandwidth and incorporating silence suppression and data techniques allows the algorithms to produce variable bit rate streams. These techniques allow data streams to be transported more efficiently by packet switched networks.

It is of foremost important that the system be designed to survive in a hostile environment. The shipboard environment is full of conditions that generate interference or are generally non-conducive to friendly circumstances. This is contrary to business environments for which most of COTS equipment has been designed.

Implementation of electronics utilizing lower power consumption also implies that lower signal strengths are required to properly operating the equipment. Without proper considerations, the lower signal strength can be overpowered by equipment in relatively close proximity with high power spurious output or radio frequency emissions. The measures applied to counter interference and protect the equipment often produce other side effects that are non-conducive to shipboard operations. For example, to counter the electromagnetic interference additional shielding may be added or fiber optic cable may be used in place of copper wiring. Adding enough shielding to adequately protect the electronics adds significant weight to the system or replacing copper wiring with fiber optic increases the level of training required for installation and repair as well as determines additional interface protocols required for proper communications.

Trunk lines provide three prominent functions. They are the primary means of inter-nodal switch communications. They provide the system with the ability to connect to shore based PBX systems. Trunk lines are a replacement item within the LAN based telephony system. In the voice over LAN system there is no dedicated switching center that requires inter-nodal connections. This elimination of the inter-nodal trunking specification also, in part, satisfies a portion of the non-blocking requirement. Other trunking requirements are simply a requirement for large bandwidth dialup connections or connections to external PBX systems. Large bandwidth point-to-point service requirements are already available within the data network and should not have additional requirements within the voice system.

The trunk line is the system interface to the public switched telephone network and other external multi-access interface requirements. The implementation of this requirement should follow the same specification of the trinary interface. One interface should be the trunk line, the second to the LAN and the third to the telephony control system.

1. Integrated Services Digital Network (ISDN)

Integrated services digital network (ISDN) is a set of communications standards allowing a single pair of wires to carry voice, video and network services and was originally intended to replace the plain old telephone system. The basic ISDN standard system never caught on for general use in the USA but did in other countries. Call setup between switches will no longer exist in trunk format. Each call will automatically originate and terminate at the predetermined location in single channel format. There is no requirement to carry a bulk or multiplexed set of channels internal to the tactical switch. Unfortunately, ISDN has become the non-standard, standard within the North American PBX manufacturing environment. ISDN terminals from one manufacture do not usually function correctly on competing manufactures' systems.

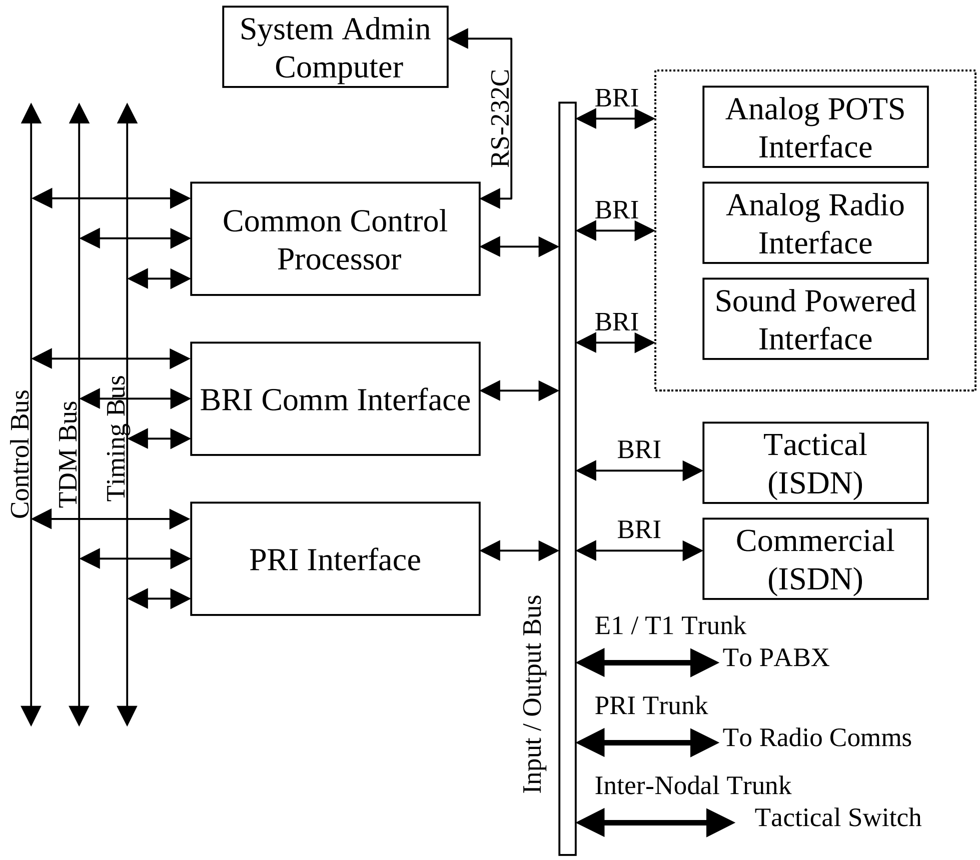

Figure 14. Bus Interface Diagram

Two of the common variations of ISDN are interfaces for shipboard use. The ISDN standard is the only standard, other than analog POTS, that is generally accepted across the industry. The primary rate interface (PRI) and basic rate interface (BRI) are the two main forms of the ISDN communications standard. This standard should be the principal standard used by the Navy for the second connection of the trinary interface (Figure 14). The PRI interface supports a point to point connection between the two PBXs and is used as the inter-nodal trunk. The BRI is a subscriber and network digital exchange allowing a full duplex transmission. The data channels are referred to as "bearer" channels and the data is referred to as the "delta" channel.

2. Basic Rate Interface (BRI) UP0

The basic rate interface (BRI) is the standard for shipboard digital (ISDN) communications. It is also the primary service offered by service providers for home use. The ISDN BRI service, has two 64k bits per second channels for carrying data or voice and 16k bits per second channel for carrying signaling and call setup. Each frame is 48 bits long and consists of two sets of 8 bit signals from both of the voice or data channels and 16 bits from signaling channel for 32 bits of payload and 16 bits for overhead. The utilization for BRI is 144 kbps. An additional 48 kbps are required for framing, synchronization and other overhead bits for a total bandwidth requirement of 192 kbps for a seventy-five percent effective utilization rate. Call setup for BRI is different from analog line because it must first verify if the receiving terminal is capable of establishing an ISDN connection, then it has to establish a digital link prior to the actual virtual call setup.

A chief advantage of using the ISDN protocol is the availability of commercially available software, commonly referred to as stacks, and electronic chip set. By utilizing the chip set in conjunction with programmable smart cards, the ISDN interface development can be significantly reduced.

3. E1 / T1 Public Automated Branch Exchange (PBAX)

The E1 is another form of the ISDN interface that is used to connect the tactical switch to the shore side communications. The E1 is the European version of the same thing. The same chip set can be used for E1 or T1. The protocols are different and that is a software implementation issue. Currently the T1 / E1 interface on the tactical switch is not employed and there is no expectation for future use.

H. BASIC PBX SOFTWARE SERVICES

1. Introduction

This software services description provides a high level functionality over view of the implementation requirements. Performance of the local area network version of the tactical communications system has to meet, at a minimum, the same performance and operational specifications as the current system. The new tactical communications LAN based tactical switch will contain additional traditional PBX services and extended PBX functionality commonly referred to as hotel services not normally found within the tactical environment. The addition of any new functionality cannot degrade the operational requirements of the existing design.

The call control processor (CCP) is the heart of operations within the traditional PBX. The CCP processes, synchronizes, and controls all operations within the tactical switch's communication. The call control process services several different functionalities of the switch. The CCP has the ability to inform the caller of specific status conditions on the progress of the call. The functionality of the CCP provides service to the terminal when the user requests call placement. When the terminal goes off-hook it has a direct, hard wire, connection into the CCP. Each network terminal is allocated a time slice within the CCP processing cycle in which all services related to that NT must be serviced. This allows the CCP to directly monitor and control all aspects of the network terminal.

In the LAN based telephony the ability to connect into the telephony server is not assumed. The telephony co-resides on the LAN with competing users that do not interface with the phone system. When a terminal on the LAN based system goes off-hook, a message has to be sent to the telephony server announcing the active status of the terminal. When the telephony server receives a terminal command, the telephony server sets up the channel connection to the server providing PBX services. Once the network connection is established the telephony server acts as a system monitor and only provides limited services to the NT.

The services that are provided by the network based telephony system have to appear to the user as if it is from a traditional telephone system. This includes the ability to connect to non-propriety network terminals in a plug-and-play environment. Because the network system is a distributive based application and not a centrally controlled processes the functionality of the traditionally based applications has to be performed in areas not under the direct control. The telephony server can only monitor performance standards and make service requests to resources not wholly owned by the tactical system.

2. Call Progression Tones

Call progression tones are services provided by the system to inform the user of specific conditions during the attempt to setup a call. Normally, the tone generator is located under the direct control of the CCP. When a call requires the service of a progression tone, the processor treats the tone as a specialized type of call. This is as simple as obtaining a time slice from the call progression tone and placing it in the time slice of the out going terminal byte.

Call progression tones are telephony system generated tones that inform the caller of significant events within the call process. With the exception of the dial tone, ring tone and ring back tone, all call progression tone events occur at the conclusion of the call attempt and continue until the call is terminated by an on-hook command of the calling party or a time out message from either the called party terminal or tone generator terminal.

Although the functionality of the call progression tone generator is the same in both the traditional and LAN based systems, to optimize efficiency the location and interface is different in the LAN based system. In the LAN based system, a call requiring call progression tones is treated as a special type of call placed in that the actual tone generator is external to the telephone server. This requires a unique signal handling where, in some cases, the signal being received by the user be generated in the call progression tone generator where the user's out going information is sent to the telephony server.

Call progression tones do not need to be generated within the telephony server, as they are with traditional call control firmware. In fact, it is advantageous if the location of the call progression tone is not a function of the server but generated from an independent locale. The location of the call progression tone generator is transparent to the user as long as the function, purpose and meaning of the tones remains the same. Because they are digital in nature, it is possible for them reside at several locations within the system without adverse effect on the system. Managing the progression tones using this philosophy would decrease the dependency on the system network by allowing a series of messaging or semaphores rather than passing the entire tone across the LAN.

There are three possible locations for the location of the call progression tones within the voice over LAN communications system. The first location is the traditional location of the call control processor. In the voice over LAN this would be within the telephony server. The telephony server functions as a best effort system that has minimum response times under maximum load conditions. This would permit the telephony server to function as a tone generator under moderate load conditions. The progression tone generator requires a dedicated bandwidth channel with minimum output requirements. Even though this is the traditional location of the call progression tones, this is not the recommended location of the call progression tone generator.

The most probable location of the progression tone generator within the voice over LAN tactical communications system is in a separate and distinct unit from the telephony server. Because tone generation is strictly a software function, this would permit a single tone generator to handle a virtually unlimited number of call progression tones thus freeing the telephony server to function strictly to process requests from the network terminal. The primary disadvantage of this placement is that any call progression tone that is not terminal would require the network terminal to send requests for service to one network location while receiving a signal tone from a different location.

A third location for the call progression tones would be within the network terminal trinary interface. This is possible because it is expected that the network terminal will have a significant level of intelligence and signal processing capability. There are considerable advantages in locating the call progression tones within the network terminal's trinary interface. By employing this configuration the call progression tones do not utilize any bandwidth of the network infrastructure. This would prevent the network terminal from having to receive and transmit signals from separate locations and would allow the telephony server to control call progression tones by simple commands. The primary disadvantage of this type of architecture would be the implementation of changes to the tone generator subsequent to deployment.

a. Dial Tone

Dial tone is a tone emanating from the user terminal that originates from the telephony server. When the calling terminal goes off-hook a message is sent to the telephony server that the terminal is requesting service. If the telephony server is available to service the terminal a dial tone is sent back to the call originator. Unlike most call processing tones the dial tone is a preamble to placing a call and the telephony server needs to be aware of any commands that are processed through the keypads. Like most call processing tones, the physical location of the dial tone generator does not need to be located within the telephony server, however, the ability to process commands would reside with the server.

Because the system co-resides on the LAN with unknown entities, there has to be a series of exchanges between the network terminal and the telephony server (Figure 15). All tactical network terminals are resources owned by the Telephony Server. The off-hook command is a preamble to a service request and prior to providing dial tone the telephony server and network terminal have to exchange a series of messages that confirm the rights and privileges of that NT to request services from the server and other entities owned by the server. If the NT does not provide the proper authentication or the class marking do not match, the services are denied.

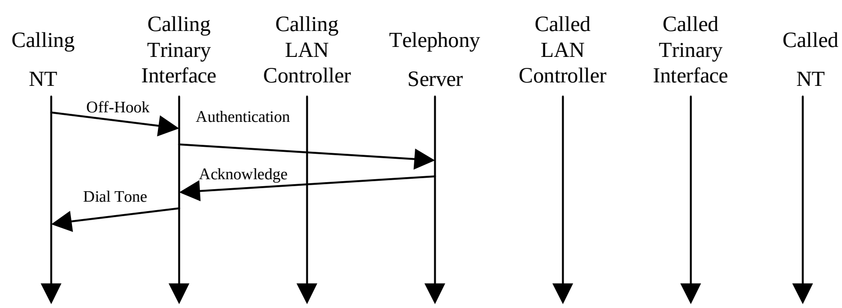

Figure 15. Call Progression Dial Tone Event Trace

b. Ring Tones

The ring tones consist of two separate tones, but are interconnected to a single function. The forward ring tone is the standard ring associated with the receiving terminal when the circuit has been completed from a calling terminal. The back ring is the tone that the calling terminal receives to indicate that circuit has been completed on the receiving terminal that is being notified on an incoming call. These ring tones generated from actions of the calling or receiving terminals originate within the CCP. Ring tones (Figure 16) are not generated for network terminals that have auto call answering such as jack boxes, net terminals, announcing systems and radios.

In the LAN based system these tones need to appear to function identical to normal BPX services. When the receiving terminal circuit is on hook the incoming channel is routed to the forward ringer. The ring back to the calling terminal is actually generated from the CCP, not the receiving terminal, to the calling terminal. When the receiving phone goes off-hook the incoming circuit is rerouted from the forward ringer completing the connection to the analog listening and speaking devices. From this point until on-hook, of either the caller or receiving terminal, the LAN acts as a wire circuit connecting the two devices.

Prior to the forward ring the telephony server has to validate the called network terminal against an internal data to confirm the rights and privileges of that NT to receive a call from a resource owned by the network terminal. These validation steps should be nearly identical to the callers prior to supplying a dial tone.

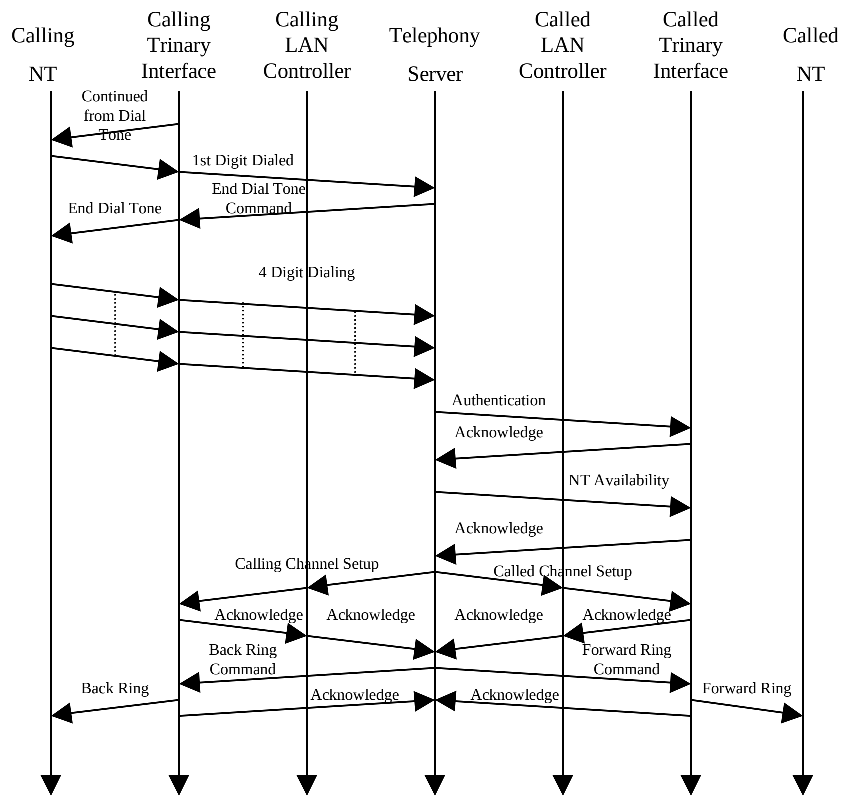

Figure 16. Ring Tone Event Trace

c. *Busy Tones*

A busy tone is used to inform the calling terminal that the unit it is attempting to call is occupied with another caller. Because the switch is non-blocking, if the caller's class marks allows, the caller can over ride the busy signal and force the call through. This is a distinguishing difference between the tactical terminal and the administrative terminal; the tactical terminal has to be able to receive and process every caller who attempts to reach the receiving terminal, up to the maximum number of terminals on the tactical system.

In the event the caller chooses to over ride the busy signal the incoming call is treated as a forward ring and the receiving terminal has several options in responding to the call. At the discretion of the receiving terminal, the incoming calling terminals can be placed in a tactical net, meet me net, local terminal net, or handled individually. The operator can choose to place the current caller(s) on hold and receive the incoming call on a separate line. The receiving terminal can allow the incoming call to become party to the current communication and, in effect, becomes a meet me net tied to the receiving caller's terminal.

d. Unavailable Tone

An unavailable tone is commonly referred to as a short busy and indicates to the caller there is some feature required for completion of the circuit that is unavailable because it is already in use or is unavailable due to outage. This signal is most common when the internal nodal trunk lines have been saturated and the caller is requesting service that requires bandwidth allocation on the trunk. If the class marking do not allow the NT non-blocked access or the call is for administrative purposes the calling NT will receive an unavailable signal from the tone generator. Because the specified requirement for the tactical communication is that it provides non-blocking services, there should not be any unavailable tone requirements because of resource non-availability.

The LAN based tactical communication system has to be a non-blocking system. There has to be enough services provided that every tactical call can get though on the first attempt. In effect, there is no unavailable tone specification for normal operation of the tactical terminal. Only on administrative terminals would a user get an unavailable tone and that would normally be associated with attempting to reserve resources outside the bounds of the ship's services.

e. Alerting Tone

The alerting tone notifies all users connected to a conference call that the circuit security level is changing. Alerting tones notify the user that the level of security of the communication has exceeded the right and or privileges of the listening terminal. This function is strictly a tactical requirement and is not required on all terminals or communications that are administrative. If the class marks on a terminal allow the terminals authorization and have ability to communicate they can increase communications to the level of the users full rights up to their classification with no additional requirements. If the terminal does not have authorization to increase security, the terminal looses the ability to continue listening to the communication. The user does not, however, loose the ability to add voice communications to the net. If the security of the conference is reduced to within the class marks of an excluded NT, an alerting tone is sounded and the terminal regains the ability for bi-directional communications.

f. Recall Dial Tone

The recall dial tone is used when the caller attempts to place a call to a network terminal and the terminal is busy. Prior to going on hook, the caller dials a predetermined number then hangs up their unit. When the called terminal becomes available the calling terminal is notified by the recall dial tone. The calling terminal exercises the option of re-calling the desired terminal by going off-hook within a specified time period. If the calling terminal exercises the option, the call is again completed to the called terminal. If the option is not exercised the recall is terminated and not further action is taken.

g. Confirmation Tone

The confirmation tone notifies a terminal user that they are being placed on or removed from a circuit that is in conference mode or is on a circuit that has the possibility of being conferenced to other network terminals. The confirmation tone is announced to all users on the network whenever any network terminal enters or leaves either a standing net or ad hoc conference call. The confirmation tone is announced as a result of a remote terminal's connection to a calling net and also announced when the remote network terminal drops out of the conference call.

h. Howler Tone

Howler tone is the announcement of alarms that is broadcast to designated areas of the ship based on the shipboard announcing system designator. This shipboard announcing system is an analog connection terminal that has no forward ring and has automatic off-hook capabilities. Once the receiving NT is off-hook a designated tone emanates from the tone generator for a specified period of time. The frequency and duration of the tones are system administrator configurable and may vary depending on the system administrator preferences. The telephone number called identifies the area of the ship to receive the broadcast. Upon termination of the tone, a one way connection is established from the calling NT to the called NT. Those terminals with the proper class marking have the right to connect to the identified number. Access by terminals without proper class marking can be accomplished by user access code entered into the touch tone pad. All other requests are terminated with a non-availability message.

i. Calibration Tones

There are two tones within the progression tone generator that are used strictly of calibration purposes. The milliwatt tone supplies a continuous tone at a specified level. The technician can then measure the quality and loss of the signal from the progression tone generator to the network terminal. The silence tone provides another means of technical support and measures the amount of noise on the transmission line between the tone generator and the network terminal. If the progression tone generator is located at a centralized location within the voice over LAN system, the calibration tones should provide the same calibration functions between the tone generator and the network terminal. If the tone generator is located within the trinary interface, to be of practical value within the voice over LAN tactical voice communication system, the calibration tones should be accessible by remote network terminals.

j. Call Progression Tone Summary

The tone generator is an integral part of the tactical telephony system. It is connected to the system through the LAN and acts as a special services network terminal. As the telephony server processes the call, signaling from the terminal and determines what call progression tone (Table 9) is required. The telephony server then connects to the port of the tone generator that provides the desired tone.

Call progressions tones consist of five classes of signals that the tone generator terminal is required to service. The first class of service, terminal tones, provide the user with the status of an attempt to connect to a terminal or conference net. The second class of tones notifies the user that they are switching to a new class of service but there are no additional call signaling services required by the telephony server. The third class of service tone required by the system notifies the user the terminal is entering a new class of service and additional call signaling services are required. The forth series of tones provides assistance for system calibration. The fifth class of tone service is shipboard alarms activated by an access code or by a network terminal with the proper class of service markings.

The tone generator allows the telephony server to process the call signaling as if the tone generator is a specialized class of network terminal. This requires the simplex communication between the network terminal, the tone generator, and the telephony server. Once the connection is made the tone generator and telephony server drops out the communication and returns only when the calling terminal goes on-hook or the tone generating terminal times out and requests termination of the communication. Under normal operating conditions the network terminals notify the telephony server when the call has been complete so that it can perform house keeping services. As a backup precaution, the telephony server continues to monitor the communications to ensure QoS and to perform maintenance procedures and to determine if there is a fault within the communication circuit.

Tone	Frequency	Cadence	Purpose
Dial Tone	350Hz-440Hz	Continuos	Ready for service.
Forward Ring Tone	440Hz-480Hz	1s On, 3s Off, Repeat	Incoming Communication.
Ringback Tone	440Hz-480Hz	1s On, 3s Off, Repeat	Waiting for call answer.
Busy Tone	480Hz–620Hz	0.5s On, 0.5s Off, Repeat.	Requested terminal busy.
Unavailable Tone	480Hz-620Hz	0.25s On, 0.25s Off, Repeat.	Service to requested NT unavailable.
Alerting Tone	1kHz	Single 0.3s to 0.5s Burst.	Change in security level of call.
Recall Dial Tone	350Hz-400Hz	0.08s - 0.12s On, 0.08s - 0.12s Off, Repeat Three Times, Dial Tone.	Previously busy NT now available.
Conference Tone	620Hz	0.25s On, Off.	Connecting to net channel.
Howler Tone	1500Hz	3s On, Off.	Connection to announcing system.
Milliwatt Tone	1kHz	Continuos At 0 dB.	Testing and calibration. Available to any terminal.
Silence Tone	None	None	Testing and calibration. Available to any terminal.

Table 9. Network Terminal Progression Tones

3. Direct Inward Dialing

Direct inward dialing is the ability of the system to connect terminal to terminal within the confines of the shipboard system. The telephony system identifies direct inboard dialing by the first digit of the dialing sequence. The actual digit employed as the first digit is programmable by the system administrator interface. When the pre-programmed digit is dialed the system then recognizes that the call is within the confines of the system and requires an additional three digits to complete the identification of the destination terminal. The digit selected as the inward preamble should comply with the standard shipboard dial plan.

4. Direct Outward Dialing

Direct outward dialing is the ability of the system to obtain a connection to an external network services without operator intervention. Direct outward dialing supports administrative services, is an administrative function and not a required feature for tactical communications support. Dialing a pre-programmed digit accesses direct outward dialing. When the pre-programmed digit is dialed the system checks to determine if an external line is available. If there are no lines available the system returns a short busy signal to the requesting terminal. If there is a line available the system reserves system resources and sets up a channel to allow communication between the terminal and trunk lines.

Direct outward dialing posses significant issues for the voice over LAN development. When communication extends beyond the confines of the control LAN environment it has to conform to communications industry established standards. This requires the protocols and signaling to back fit requirements even though there are standards that are better suited for today communications needs.

Additional hotel services are required to support direct outward dialing. The system has to have the ability to allow the user to place credit card calls, collect calls, and special billing cards without the assistance of an operator. This is a particular challenge within the LAN telephony system. Commands over the telephony system are given and received in digital format while commands given in a commercial PBX are in analog format. To meet both requirements the LAN based system must have a point of demarcation where a command, such as a key press from a tactical terminal, is converted into the analog equivalent acceptable to the external PBX.

The system also needs to provide for call cost accounting capabilities. Whenever a call is placed between terminals within the switch or charges are accepted from calls external to the switch, the system has to have the ability to track and record the charges incurred by terminals. These charges may be outward shoreline connects as well as internal satellite that are ship to shore connections.

5. Automatic Route Selection

The software has to provide automatic route selection across the local area network. The system needs to provide for adaptive routing procedures to compensate for network traffic pattern changes, circuit availability, or partial system failures. There are two primary reasons for requiring the automatic routing of signals within the LAN environment. The primary reason is circuit failure protection and the alternate reason is load balancing. Load balancing is not a significant concern within the Navy environment, because the total bandwidth allocation is significantly greater the utilization requirements. In the future Navy requirements and in the commercial environment load balancing is a significant concern. Most existing LAN technologies support automatic re-routing of signals upon the failure of a circuit. Employing one of these topologies will greatly reduce the development overhead required to convert tactical communication to the network environment. Care needs to be taken in ensuring that the desired LAN can meet the automatic circuit rerouting without the disruption of service on any established circuits or the system monitoring by the telephony server.

6. Net (Conference) Calling

Net calling in tactical communications operates similar to conference calling in PBX operations. In tactical communications there are three types of net calling. In tactical operations the first two net calling schemes, the tactical command net and meet-me net, are identical in operation. The command net is an assigned station for the terminal or operator to dial into, depending on the status or alarm condition of the ship. The met me net operates in identical fashion as the command net, but participation is ad-hoc and terminals or operators are not pre-assigned and enter the net voluntary. The third type of net calling function in the same fashion as tradition PBX operations where a user can be added by other another user already on the net are then removed when their connection is terminated to the connected NT. The net ceases to exist when all calls are terminated.

Because of the ability of any tactical terminal to connect to any net, multiple nets, any terminal and multiple terminals, it is possible for one terminal to render the entire tactical telephone system inoperable. To prevent improper operation of a network terminal, access or classmarks (privileges and restrictions) can be pre-assigned to any terminal limiting the functionality of the specified terminal. For some integrated terminal interfaces the user may just want to monitor what is going on but not actively talk on the circuit. At other times the user may want to talk while several other users only listen.

Both the tactical and met-me nets are always operational, whether or not there are terminals currently dialed into it. Any tactical terminal, with the proper class marks, can call into the net and establish a connection at any time. Any user may leave the net without disruption of service to any other terminals currently on the net. Conceptually, the net is a perpetual party line where terminals may enter or leave at will without loss of signal.

7. Sound Powered Nets

The primary user of the two primary shipboard nets is the sound powered telephone (Figure 17). The sound power phone is a specialized tactical terminal. The sound powered net is a string of one to many sound powered phones that have the ability to communicate across a common cable without external power. Terminals connected to the LAN based tactical telephony server have the ability to call into the sound powered net by dialing a number associated with the net. The number of network terminals from the tactical switch dial into the net of the sound power telephone is unlimited. The sound powered net does not have the ability to dial out and is permanently assigned to a tactical net. In essence it is a one way connection where sound powered phones have the ability to instantaneously connect to other sound powered phone on the net and where other phones have to dial in to connect.

In commercial voice communications a bridge circuit is used to accomplish the net feature. In the tactical voice over LAN a similar feature needs to be utilized. There are specified phone numbers to dial when attempting to connect to a specified net. Once a terminal has dialed into the net their incoming signal is summed with all other signals attached to the designated number and sent out to all connected terminals.

Although the telephony server could function in this capacity, it is not recommended that this approach be utilized. The desired approach would be to add net servers to the LAN that function as a specialized network terminal or conference bridge. This would permit expansion of the net bridges by the simple addition of servers to the LAN and would free the telephony servers of additional responsibility. Because of the low demand utilization on the net server, it would be expected that more than one server could be incorporated into any single unit.

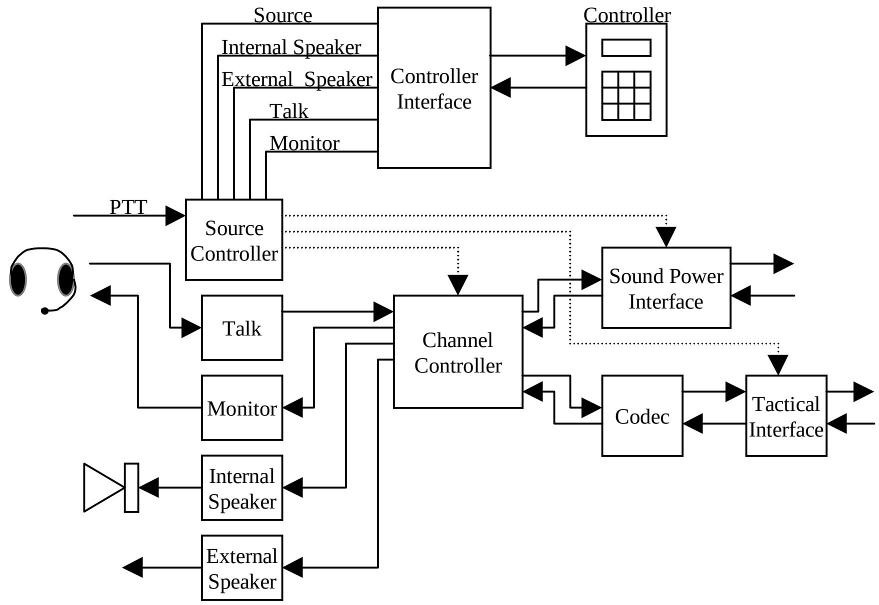

Figure 17. Integrated Terminal Equipment Block Diagram

a. Abbreviated And Speed Calling

In the current tactical system speed dialing is referenced as abbreviated addressing or dialing. Speed dialing and abbreviated dialing appears functionally similar to the operator but its implementation is not functionally the same. Speed calling is the ability to place a call to a tactical net with a limited button press. This limited sequence of dialed digits invokes a full dialing sequence that is sent out to a call control processor. Abbreviated dialing consisted of ability to walk down a dialing tree and have the call completed based on the termination of the tree walk. The tree termination may be as limited as a single digit or as much as four digits, which can access all internal network terminals. Both functions are required in the tactical telephony system. The implementation of speed dialing will most likely occur within the tactical terminal while abbreviated dialing will functionally take place in the telephony server.

Individual speed calling has the ability of the user to program the local terminal with a redefined set of phone numbers according to their needs and desires.

Individual speed calling functionality is located within the local terminal and is not controlled by any external means. The operator predefines a number then stores that number in allocation within the terminal's memory. To recall the number the operator selects the recall function and the desired memory location to be recalled.

8. 18-Digit International Number Routing

Because terminals are used for administrative purposes as well as tactical, a terminal with the proper class markings is able to access trunk lines outside the limits of the tactical communication system. Eighteen digit dialing is the ability of a terminal to access international numbering plans. This ability is an administrative function not associated with tactical communications. This feature is not normally associated with underway operations and in used primarily when the ship is located pier side and the system is interconnected to shore communication facilities.

The eighteen digit dialing plan is accessed by dialing a specified prefix, such as the digit 9, which then connects the caller to a PBX external to the ship which has international dialing capability. The caller is connected to an external PBX via common interface format and protocols. This not only requires the tactical voice system to have external capabilities, but also requires it to maintain a common interface standards and protocols. In the voice over LAN effort this will require a specialized that interfaces to the external PBX.

For the voice over LAN to be a practical solution the internal system formats and protocols should match common industry standards so the extended format appears as an extension to the internal system. In the event a common protocol standard is not achievable within the tactical voice over LAN system, an interface should be provided to the commercial PBX that converts internal signals and protocols to commercially acceptable formats.

9. Caller Identification

Caller identification is the ability of the called terminal to be able to identify a calling terminal's phone number and terminal station personal identifier prior to answering (going off-hook) the call. The system administrator has the ability to assign caller identification according to a predetermined calling plan. Each telephony server maintains a configuration containing the system's entire caller identification database. The data is updated using the system administrator's interface. When the database is modified within one telephony server it is then updated in all remaining telephony servers. When telephony server comes on line for the first time it is also updated with the current caller identification database.

10. Call Waiting

Call waiting is the ability of the receiving terminal, that is already off-hook, to know there are one or more callers attempting to establish a connection. The called terminal will have the ability to perform caller identification of any incoming calls that are waiting to be established and select the desired calling terminal and place the existing connect on hold. Because of the non-blocking requirement, all tactical network terminals have to be able to accept all incoming calls. Call waiting adds to the ability of non-tactical phone to accept at least one additional call before blocking is implemented.

The system establishes call waiting the same way it sets up a normal communications channel. Because the terminal is already off-hook the receiving terminal does not send back an off-hook command to the sending terminal. The incoming call rings back as if the receiving terminal was on-hook. Rather than producing a normal forward ring the receiving terminal produces an audible indicator on the receiving terminal indicating there is a call attempting to connect. The audible indicator is repeated at a given interval until the caller goes on-hook or the called terminal places the incoming call on hold and picks up the incoming call. As with any call, either terminal has the ability to place the other party on hold, or to any net they are currently connected.

11. Call Forwarding

Call forwarding automatically transfers any incoming calls from a receiving terminal to another pre-selected receiving terminal. The calling terminal cannot abort the transfer other than to terminate the call. Implementation of this feature could take place in receiving terminal, but this would in effect double the number of steps to establish communication. The desired location for implementation would be in the telephony server. The requesting network terminal would dial a predetermined phone number where the server would identify the calling unit and the user could selected the desired forwarding number. Cancellation of the service would be accomplished in a similar manner. An operator would only be allowed to implement and cancel call forwarding from the calling terminal.

12. Call Hold

Call hold allows network terminal with the class of service to place a calling party in an inactive communication status. The network terminal that activates call hold deactivates all speaker and microphone capabilities. Call hold is activated within the network terminal and has no effect on the network circuitry. No interaction is required from the telephony server or another device external to the activating network terminal. All connections remain in effect and no external operations are required. While in an inactive status either party can perform any other function of the terminal.

13. Call Park

Call park is similar in features to call transfer in that the answering party of an established call can park the calling terminal at a third party. The third party terminal then has the option to retrieve the parked call with a predetermined dial code. If the third party does not retrieve the call within a specified time, the calling terminal then rings the terminal that placed them on call park. If the third party does not retrieve the call within a specified period of time the parked call rings back to answering party.

The implementation of call park requires the services of the telephony server and the parking terminal for successful completion. The services of call park should be preprogrammed into the terminal. The operator selects the caller to be placed on call park, then the call park feature is selected, the destination of the call park is selected and finally the call is transferred. The forward ring, time out and ring back to the transferring party are automatic and controlled within the transferring terminal. The telephony server does not require any knowledge of the implemented actions until the call is completed at the forward ring. At this time the telephony server establishes a connection between the transferred and forward called terminal and disestablishes the original channel.

14. Call Transfer

Call transfer is the ability of a network terminal to transfer any call, in its entirety, to a third terminal. The telephony server uses the class of service and class marks of the terminal being transferred in determining the caller’s rights and privileges in establishing the new connection. Call transfer is setup by the telephony server with the terminal initiating the transfer acting as an agent for the transferring terminal. Once the call is completed to the third party across the network, the telephony server disconnects and tears down the circuit between the original parties.

15. Call Groups

Call groups is the ability of the network terminal to perform net conferencing. There are three types of net conferencing located within the tactical environment. The first two types of call groups function exactly the same operationally; it is how they are employed that differs. The first two types, the tactical net and meet me net, are conference call bridges that are always active and any terminals with the system that has adequate class of service and has the class marks can access the call groups at any time. The tactical net is used strictly for tactical purposes and shipboard management determines its use. The meet-me-net is used for conference calling that is not tactical in nature. The meet me net, like the tactical net, is always in existence and can have zero to many network terminals connected. The third type of conference calling is established only when a third network terminal is added to an existing call. This conference call is only in existence for the duration of the call. The conference is terminated as soon as the last network terminal terminates its connection with the last network terminal within the net.

16. Camp-On

Camp on is the ability of a calling terminal that encounters a busy terminal to be notified when the called terminal is on-hook and the circuit is cleared and ready to receive an incoming call. The calling terminal can initiate camp-on when it encounters a busy signal by informing the telephony server that it desires to be notified when the called terminal is available to receive calls. Once the desired circuit is clear the telephony server rings back the caller. When the caller goes off hook the operator has the option to establish a connection to the requested terminal or cancel the camp-on feature by going on-hook.

Camp-on is implemented in the telephony sever and is based on the classmarks of the network terminal. A tactical network terminal with non-blocking class of service cannot have camp-on based on classmarks. When camp-on is activated the network terminal notifies the telephone server of the request to be notified when the circuit of the network terminal becomes available.

17. Messaging

Messaging is the ability to communicate between terminals using an alphanumeric based store and forward procedure. The calling user enters a message using the terminal's data entry device and addresses the message to the desired receiving terminal. The message is routed through the telephony server as a designated source and then forwarded to the receiving terminal. The receiving caller can then review the message at their discretion. Messaging does not require the system to establish a dedicated circuit and is provided using the best effort of the system.

18. Voice Mail

Voice mail is the ability of the caller to leave voice messages, which the called terminal can retrieve at a later time and date. Voice mail is activated when the call terminal is busy or forward rings a predetermined number of times and the called terminal does not go off-hook. Voice mail is pre-selected and monitored in the telephony server. When the telephony server determines the calling terminal has reached the criteria, the incoming call is rerouted to a predetermined smart recording device. The telephony server notifies the smart recording device of the called terminals identification and the messaging device prepares call handling procedures unique to the called and caller terminals.

If the caller leaves a message the smart recording device notifies the called terminal that there is a message in the terminal's message box. The called person can then retrieve the message from any terminal using the called terminal's line number and a password.

19. Push To Talk (PPT)

Push to talk (PPT) is a feature not normally found in commercial telephone systems, but is essential in the tactical communications environment. The PTT feature supports various radio nets, internal nets, intercom, interphone, and radio communications. The PTT feature is employed in simplex communication and where any background noise, from multiple network terminals, interferes with the ability of the listener to clearly hear the communications. The talker (sender) of the communications activates PTT by depressing a button located on the transmitting network terminal. For radio communications, the system manages the audio within the system by the internal speaker carrying speech on a plain circuit. PTT signaling, is not only required at the transmitting NT but has to be passed through to the entire circuit to the receiving network terminal.

The telephone server plays an active role in push to talk by determining if a network terminal has the push to talk class of service and to determine if the proper classmarks are in place. Push to talk is activated and de-activated in the network terminal's transmitting channel and does not interfere with the receiving channel, in bi-directional communications.

20. Special Connection Terminals

Special terminal connections are those connections that require an interface unique to a terminal type. The circuit may be a one to one connection, one to many connections or a device with multiple line interfaces where the incoming caller is routed to the next available circuit. For the operator to communicate on the circuit they must activate the push-to-talk feature. The primary capability of special connection terminals is the ability of the receiving connection to complete the call without operator intervention. If the circuit is available, the receiving connection automatically forwards the off-hook command to the telephony server.

21. Shipboard Alarms

Network terminals with the proper class of service and with the proper applied classmark will be able to sound shipboard alarms, such as general quarters, by activating a key feature or entering a predetermined dial key code. Alarms are activated via a special connection to the shipboard announcing system. The alarm sounds based on the activation of the circuit and ceases when the communication is terminated. The alarm will momentarily pause when the PTT is activated, allowing simplex communications to special connection, and resume when the PTT is deactivated.

22. Class Of Service

Each terminal type connected to the tactical LAN based communications system has certain capabilities and limitations. The class of service for each terminal type identifies these capabilities and limitations. When a terminal attempts a connection to a terminal that is of a different class of service the system has to designate a communications protocol prior to opening a channel for communications. It is the responsibility of the telephony server to negotiate between the network terminals until a mutually acceptable protocol is found. Most likely it will be the responsibility of the higher class of service network terminal to convert to the protocol of the lower class of service device. Not all communications are possible from all network terminals. For tactical communications the tactical terminals will have the highest class of service allowed by the system. Administrative terminals and special connections will have class of service limitations.

A primary function of the class of service is system security. Each network resource has the ability to uniquely identify itself, its location on the network, and its class of service to the telephony server. The telephony server compares the unit's identification and the class of service sent to it against a database, preprogrammed by the system administrator, prior to allowing the NT to communicate across the network. This verification and validation process allows only authorized network terminals to access the assigned network channels and utilize system resources. Any attempt to circumvent the process and access system resources, including network terminals, without following the proper protocols would alarm the system administrator that improper activities were taking place.

23. Classmarks

Each type of terminal located within the confines of the telephony system has certain rights and privileges within the confines of the telephony system. Every network terminal on the tactical telephony system contains classmarks. Classmarks provides a means for the system administrator to assign rights and privileges for the operation of any given network device that further limit the class of service. A network terminal may have the ability to perform given telephony functions, but may not have the rights assigned to it to use those functions. There are certain privileges, such as overriding an existing connections and designated phone numbers, that have limited access rights. The class of service is assigned by the system administrator through the system administrator interface and is maintained within the telephony server database.

24. Classified Communications

The system has to allow a network terminal with the proper class of service and classmarks to conduct unclassified and classified comminations concurrently. Classified communications is the ability to interchange information that has been classified by the United States government to be sensitive enough in nature that if it were to be acquired by uncleared sources could place the warfighter in jeopardy. The ability to pass classified information by electronic means has to be approved by a government agency with the authorization to make such a determination. At no time can an operator pass classified information by electronic means until the authorizing authority has approved the system in its entirety.

25. Sensitive Communications

The system has to allow a network terminal with the proper class of service and classmarks to conduct sensitive and non-sensitive communications concurrently. Sensitive information usually reveals knowledge that is not classified in nature, but should not be provided to users of the system who are not authorized to know such information. The ability to pass sensitive information by electronic means does not have to be approved by an authorized government agency and only needs to meet the specification of limiting access to sensitive information.

I. SHIPBOARD DATA NETWORKS

1. Introduction

The shipboard local area network (LAN) is a resource not commonly available for real time dynamic allocation of the tactical telephony system. The network is a shipboard resource that needs to be evaluated to determine if it can provide sufficient bandwidth, throughput requirements and quality of service abilities to handle minimum tactical telephony system requirements. This profile of networks is limited in scope and is not intended to eliminate specific types of technologies, but it is designed to suggest what type industry standards technologies that do lend themselves to tactical level communications.

2. Elements Affecting Shipboard Networks

Shipboard broadband networks are a relatively new feature and, at this time, bandwidth allocation within the confines of the ship does not present a significant limitation. This is further amplified because most major industry standard network backbones can be installed with adequate bandwidth that all applications, regardless of bandwidth utilization or data transfer technology type, can be accommodated. As the network becomes a standard shipboard feature it is fully expected that utilization will be maximized to its full potential bandwidth and utilization may be more of a concern.

Migrating the IC requirements to the data network requires the sharing of resources by users other than the tactical telephony system. The LAN is shared by differing data transfer protocols and is required to co-reside with classified communications. In the circuit, the network is not a resource owned by the telephony system. For successful integration of the tactical voice system into the data shipboard network a system of rights, privileges and specified minimum level of performance needs to be established.

Bandwidth on demand is a technology of emerging network backbones. Although bandwidth on demand is a beneficial technology where traffic is bursty in nature, voice communication circuits must be designed to handle minimum sustainable levels of bandwidth utilization. During “busy hour” the network must be able to handle all critical circuits of both voice and data without delays or loss of QoS.

In data networks, a best effort requirement is adequate to meet the minimum requirements By introducing buffering and small delays in communications, total utilization of a channel can exceed the maximum limits for short periods of time; the bandwidth can exceed peak capability and still function acceptably. In voice communications, if the limit of bandwidth utilization is exceeded there can be a significant loss in the quality of service. If the loss is severe, then communications becomes unintelligible. To prevent QoS loss the total utilization of the network bandwidth must never exceed the ability of the voice communications system to meet the minimum bandwidth requirement of any single channel.

Bandwidth on demand becomes an effective technology if the system is not capable of handling the maximum utilization of both tactical and administrative communication requirements. This does not pose a significant problem when considering that most current backbone technologies being installed aboard ships have OC-12 bandwidth rates with sustainable utilization at rates of ninety percent or better. This becomes critical, however, with LAN technologies with limited bandwidth in the outlying areas.

Bandwidth on demand becomes practical if there is the ability to prioritize communications and allocate or de-allocate bandwidth accordingly. This requires the development of a LAN wide resource allocation system. This over arching system would have to contain information of all critical systems and their priority within the communication environment. As the utilization of the bandwidth reached the maximum potential, the system would reallocate bandwidth according to a predetermined seniority strategy. Systems unknown to the administrator would be allowed to utilize bandwidth on an as available bases, but would be the first to be eliminated as resources became critical.

3. Shipboard Local Area Networks

Because the shipboard network is not a resource owned or maintained by the tactical voice system, horizontal integration of two competing technologies needs to occur. No LAN based technology should be eliminated strictly based on topography or protocol type. Any network that can meet minimum requirements should be considered. The development should have the ability to eventually utilize any suitable type of network protocol or topography.

Some technologies better lend themselves to this type of development. For the preliminary prototype a best suited technology needs to be chosen. The selection of the desired system should include shipboard availability, overall costs, and ability to support emerging technologies as well as the capacity to accomplish the primary objective. To better make this selection a basic understanding is required of those technologies that better lend themselves to this type of development and are those that are the most probable choices.

a. Carrier Sense Multiple Access / Collision Detection (CSMA/CD)

Ethernet is the primary local area network used within industry and within the Navy. The specifications and rights to build Ethernet technology are publicly available creating a large market of competitively priced equipment. The Institute of Electrical and Electronics Engineers (IEEE) published the original standard under the title “Carrier Sense Multiple Access with Collision Detection (CSMA/CD)”and is adopted by the International Organization for Standards (ISO) making it a worldwide networking standard.

Bandwidth	Overhead Per Packet	Minimum Packet Size	Maximum Packet Size
10M Bits	18 Bytes	64 Bytes	1518 Bytes

Table 10. Ethernet Frame

The most widely used version of Ethernet is ten megabits per second (Table 10). It is also available at bandwidths of one hundred and one thousand megabits per second. All stations are connected to a shared signaling system. Each station operates independently of all other stations and there is no central controller. Signals are transmitted serially over the shared signal channel to every attached station. To send data a station must listen to the channel and when the channel is idle the station transmits data in the form of a frame or packet. Each station with data to transmit must contend equally for the next transmission opportunity.

The logical topology of the Ethernet provides a single channel or bus that carries the signal to all stations. Multiple segments can be linked together to form a larger LAN using a repeater. Multiple segments grow as a non-root branching tree and each media segment is an individual branch of the complete signal segment. The media segments may be connected in a physical star pattern, with multiple segments attached to the repeater and the logical topology is still that of a single channel that carries signals to all stations. Segments can never be connected in a loop. The total set of segments must meet the round trip timing specifications as if it was a single LAN.

	Transfer Rate	Packets (Minimum Packet Size)	Packets (Maximum Packet Size)
Voice	64K Bytes / Second	1391.3 Packets / Second (46 Byte Payload)	42.12 Packets / Second (1500 Byte Payload)

Table 11. Combining Simplex Voice on 10 Mbit Ethernet

If more than one station attempts to transmit on the channel at any give time, the signals are said to have collided. The stations are notified of this event and reschedule their transmission at a randomly chosen microsecond interval. This keeps the stations from again attempting to transmit at the same time. In the unlikely event the collisions for a given packet transmission exceeds 16 consecutive collisions the interface will discard the packet.

Hubs may also be used to extend the operational capabilities of each LAN segment. Connecting the Ethernet LAN together with hubs allows each segment of the LAN to operate as if it was a separate and distinct LAN where the timing requirements are only applicable to the individual segment and not to the LAN as a whole.

Voice Transfer Rate - Simplex	Ethernet Bandwidth	Bandwidth Utilization (Minimum Packet Size)	Bandwidth Utilization (Maximum Packet Size)
64K Byte/ s	10M Bit / S	7.12%	5.11%

Table 12. Simplex Voice Channel over Ethernet Efficiency

Ethernet LANs operates as a best effort data delivery system (Table 11) while affording each user equal opportunity to gain access to the network. Ethernet technology does not have the ability to prioritize the users, deny services to unknown users or to users of low priority, guaranteeing a minimum level of performance. In addition, the typical maximum utilization of this type of technology is usually no greater than forty percent of the total bandwidth (Table 12). For these reasons it is not recommended that CSMA/CD be developed for the tactical aspects of the communications system.

b. Token Ring

Token ring is protocol architecture developed by International Business Machine (IBM) in the 1970s. It is a star wired ring topology having token passing as its network access method using a baseband transmission technique operating at 4 megabits per second (Table 13). It is known as a deterministic method of access since a station can only transmit data when the token is available to it. Once a station releases the token the next station in the channel grabs it. No collisions can occur because only the station with the token can transmit. Each station grabs the token when it is passed whether they have data to transmit or not.

Token	Header	Payload	Frame Size	Efficiency
3 Bytes	11 Bytes	0-4500 Bytes	14-4,514 Bytes	0-99.6%

Table 13. Token Ring Utilization

Token ring transmission is a relatively efficient method of transmitting data when compared to the bus topology of Ethernet. Precise timing of how long a frame takes to travel the ring can be determined by the number of stations in the circuit, the propagation delay between the stations (Table 14), the amount of data to be transmitted and the number of hops the receiving station is from the transmitting station.

Propagation	Analog Voice Data Payload	Overhead	Network Terminal Channels
4 Mbit/sec	64 Kbit/sec	112 bits/frame	60 Duplex

Table 14. Analog Voice Over Token Ring

Token ring is not a common network found within the military or shipboard environment (Table 15). It is significantly more costly to purchase and maintenance personnel have to have a higher degree of expertise to properly operate the system. Token ring is more prone to total system failure occurring from minor station faults. If there is a minor problem self-recovery can take up to several seconds. For major malfunctions the ability of the system to have build in hot swappable back up is also harder to implement.

Propagation	ISDN Voice Data Payload	Overhead	Network Terminal Channels
4 Mbit/sec	144 Kbit/sec	112 bits/frame	27 Duplex

Table 15. ISDN Channel Over Token Ring

J. SHIPBOARD DATA BACKBONES

1. Asynchronous Transfer Mode (ATM)

Asynchronous Transfer Mode (Table 16) is the emerging technology of choice for shipboard network infrastructures. Similar to other data transfer technologies, ATM is a variable bit rate (VBR) service offering best effort to data transmissions. ATM offers several features conducive to voice communications not found in other data transfer technologies. ATM also has the ability to setup virtual constant bit rate (CBR) circuits that are in existence only for the duration of the signal. ATM offers bandwidth on demand. This allows bandwidth that would normally reserved for specified channel, when not in use, to be dynamically re-allocated to channels requiring momentary higher bandwidth. Because of the multi-users requires and that at any given moment voice communication utilization, bandwidth on demand and constant bit rate allocation are two features that allow ATM services to dominated the prevailing market.

Cell Size	Header	Payload	Efficiency
53 Bytes	5 Bytes	48 Bytes	90.6%

Table 16. ATM Cell Structure

ATM technology has a unique feature that lends itself to the tactical aspect of the voice communication is the ability of the ATM circuit to “heal” reroute and itself in the event of circuit failure. This is a primary required feature in the tactical voice communication system that is built in as a standard to the ATM design specifications. Maximizing this advantage would severely reduce the redundancy requirement.

Even though it is referred to as ATM technology and switches, there is a minimum of three different major subsystems required to have a functioning ATM network. ATM is the technology of the switch. This is the technology that routes the information from the switches input ports to the output ports within the switch. Synchronous optical network (SONET) (Table 17) is the technology that interconnects all the switches. As in ISDN, the actual protocols going to the end terminal equipment are different that those connecting the switches point to point. The third technology within the network is what is required to deliver the data the end terminal equipment. This technology may be ATM to the desktop, but most likely will be more telephony in nature, in order to meet the COTS terminal requirements.

Frame Size	Overhead	Payload	Bandwidth	Frame Speed	Efficiency
810 Bytes	28 Bytes	782 Bytes	51.84M bit/s	125µs	96.5%

Table 17. SONET STS-1 Structure For ATM To The Network Terminal

ATM has several inherent benefits and that help ensure quality of service (QoS) and make the service a prime candidate for network services on the tactical voice communications environment. The service has dynamic bandwidth utilization via multimedia services. The service has the ability to dynamically prioritize utilization of classes of signals within each channel. ATM can dedicate bandwidth for constant bit rate (CBR) service. The fourth feature that makes ATM a popular choice for tactical communications is the automatic rerouting of signals in the event of equipment failure or circuit loss. By incorporating these inherent features of ATM into the implementation of the system the overall system reliability can be increased while reducing the cost and time of the development.

Originally ATM proponents claimed it to be the end-all network solution. There are significant drawbacks employing ATM technologies in telephony developments. The technology is not mature enough or the standards accepted industry wide, so the equipment from different manufactures has standardized on slightly differing interfaces. This makes the technology a single vendor solution and increases the probability of incompatibly with emerging enhancements. Because ATM communications conventions is many to many it can be difficult to scale the network. ATM is significantly more complicated to design, install and maintain than gigabit Ethernet or even FDDI. ATM is costly and complex. (Gigabit Ethernet is a less costly and simpler alternative.) This makes the technology complex and more difficult to maintain, requiring a much higher degree of technical sophistication.

2. Fiber Distribution Data Interface (FDDI)

FDDI is a technology (Table 18) that has operating features similar to token ring. The network has two one hundred megabit per second counter rotating rings. Only one ring, the anti-clockwise ring, is active unless a fault occurs. A token is passed from one node to the next and only the possessor of the node can communicate on the ring at any one time. The times for servicing each of outlying nodes can be precisely predicted thus ensuring a minimum service requirement. If the primary ring fails the secondary ring automatically takes over communications. All nodes read the token to determine if they are the recipients. If node is the recipient, it extracts the data. If a frame cycles back to sender, the originating station strips the frame and releases the token.

The primary advantage of FDDI technology is that it is proven technology. The cost of implementation is relatively low and the solution can cross multi-vendors. The token ring holding timer determines how many frames can be transmitted before it has to move on to the next station resulting in a more precise knowledge of the time it will require to transmit any data. The primary disadvantages of FDDI is its inability to self heal if the primary and secondary ring is broken. There is not an ability for the network to self-monitor and control the problems within the structure.

Token	Header	Payload	Frame Size	Efficiency
11 Bytes	17 Bytes	0-4478 Bytes	28-4,506 Bytes	0-99.3%

Table 18. FDDI Cell Structure

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V. COST OF OWNERSHIP

A. INTRODUCTION

During the early development of communications the military was a driving force behind the development of certain types of technology. These technologies were developed at the expense of the government then converted to civilian use. Subsequently, commercial development has vastly surpassed the military in many arenas. Commercial of the shelf (COTS) equipment is being deployed with better features and manufactured with a higher degree of reliability than can be afforded by the military within the same development window.

It is more economical to deploy commercial systems than it is to develop in-house systems unique to the Navy. The fundamental obstacle is that COTS systems do not meet the total needs of the military tactical communication environment. The primary cause is the limited market presented within the Navy. Research and development costs can be exceptionally high to meet unique needs. The vendor has to be able to market enough of these systems to cover their development costs. With the limited market presents of the Navy, there is little opportunity for the developer to recover these costs.

Because developing systems, such as the tactical voice communication system, are expensive the objective of the development should be to maximize the utilization of COTS equipment when ever possible and only develop new systems and sub-systems when the functional needs of the shipboard requirements do not already exist within the commercial environment. The resulting hybrid system would still be unique to the Navy, but by following accepted commercial standards with in-house developed sub-components there is a higher probability that commercial manufactures will provide these services as the demand for niche market increases. Using this methodology for the project will have the least impact on the Navy’s development dollar.

B. SYSTEM RELIABILITY

System reliability is not only the ability to properly function for extended periods of time without failure, but also to be able to properly function in an adverse or hostile environment. In the event of system failure, the failsafe procedures also have to provide for an orderly sequence of events in the instance of component, sub-system or entire system failure. This ability is the primary responsibility of the software that performs diagnostics within the total system and executes hot cut over procedures with no disruption in service. System reliability, as it relates to software engineering, not only concerns itself with the ability of the firmware to function correctly over time and in a myriad of different possible situations, but needs to provide system redundancy and backup in such a manner that no single point of failure causes a disruption in service.

C. SOFTWARE SYSTEM MANAGEMENT

In the shipboard tactical system there are several principal areas of system management concern. The first area is routine house keeping services. This is the ability for the shipboard operator to tailor the system to the specific, day to day, needs of each individual ship. The second area of software system management is the ability to maintain configuration control over an entire ship class. Configuration control is a software function usually maintained at a depot level maintenance where over arching ship to ship configuration management can be maintained. The third area of software management is the ability to sustain life cycle maintenance control of any application or firmware software. Another area of primary concern, not normally associated with self-contained systems, is the ability to maintain configuration control over non-native systems used, but not owned, by tactical voice communications. All users of shared systems have certain rights to utilize features systems within the shared systems. Any system that is modified has to be identified for use by all systems that utilize its resources.

It should be expected that the annual costs for software system management would start at approximately twenty percent of the total system development costs. As the software becomes more stable, annual system management cost will reduce to approximately ten percent of the total development costs. There will come a point of diminishing returns, when the cost to support software maintenance will far exceed the amount of productivity. Historically, it requires a minimum of one million dollars annually to maintain a minimum level of effort support for this type of development.

D. SYSTEM MAINTANCE

The system has to be developed in such a way that it includes the ability of the system to self-monitor, self-diagnose, hot cut over, and report all system failures to the level require correcting the problem. Self-diagnostics has to have the ability to detect and report in excess on ninety-eight percent of the problems of all software and hardware components. Because of the modular design of the system, there still in a small percentage of the problems that the system will not be able to report. Most of these repairs will be outside the scope of the shipboard technician and will require depot level repairs.

The system has to be designed in such a way that no single failure of a component or printed circuit board will cause the system to lose any functional capacity. Part of the system design requirements is to have at least one hot cut over capability per critical system per switch. This means that two failures of the same component or board type on the total system would not show any dedication in performance.

Because the system's built in redundancy there is inherent spare parts built into the system. The probability of three failures of the same nature within a limited window of performance is extremely remote. In the event of three or more failures of a given capability, and because of the conservative nature of shipboard tactical systems, the ship would be required to carry replacements that can be hot inserted and become fully functional without administrator intervention.

E. COST

The Navy approves for development only those systems that provide a significant increase in capability or reduction in overall costs. Interior voice communications developments are done with short term financial objectives and expending minimal development costs. Significant cost savings in the short term have to be demonstrated prior to approval of development. True cost savings may not be realized because of the inability to accurately predict actual development costs over the lifetime of the system. The problem is further compounded when development agencies low ball development costs or inflate estimated savings to gain favorable reviews in the approval process. These factors need to be taken into account when considering the system's overall lifecycle costs.

After market procurements can unexpectedly drive up a system's over all costs. The interior voice system combats this serious cost increase by purchasing a significant amount of spare parts during the initial system procurement cycle. This type of procurement assumes that the system will be in service in its initial state for a specified period of time. If the system requires upgrading or maturity of the market surpasses the technology of the current development then the system can go into obsolescence prematurely. The inclusion of two different types of technologies with associated specialized developments also increases the risk of untimely obsolescence.

The ability to combine legacy network terminals with newer LAN technologies gives greater flexibility in configuring the system to meet current needs while providing a clear migration path for the future. This incremental migration capability is provided with reduced additional up front cost while supporting the system throughout the system's life cycle. The additional development cost are incurred if there are no existing COTS systems that can meet the minimum requirements or the existing system does not meet the additional burden placed on it by the combined services. There can be significant amount of risk, but this is usually occurs in the front end of the development and the risk is gradually eliminated as the system evolves.

When ever possible, the government should not be in the business of developing interior voice communication systems or subsystems. The current state-of-the-art is already far beyond what can be developed with the government's limited funding, time and resources. By utilizing COTS components the cost of development is placed on the manufacturer. Although the manufacture's development costs are amortized into the sales price of the unit, with limited deployed systems within the Navy, the increased unit price is still substantially less than the total development costs if the Navy were to attempt to develop the same system.

Support personnel is another significant cost consideration in the design elements of a shipboard system. Shipboard labor is limited to begin with, and over the lifecycle of a system a few hours of service requirement can accumulate into a significant amount. By training shipboard personnel only in system administration functions and limiting the technical service requirements to the highest level possible, the Navy can extensively eliminate the need for highly trained service personnel.

There will still be a requirement to maintain highly qualified depot level staff of trained personnel to perform service requirements beyond the scope of shipboard maintenance. It is impracticable to maintain a highly qualified staff of shipboard maintenance personnel to repair a system that will require service only an average of every couple of years. If an element within the system fails, the diagnostics capability has to be able to have self-correction or isolation feature automatically cut in a backup. The system should identify down to the lowest replaceable unit for which the shipboard technician should have spare parts and training to replace. In the event the system is not able to accurately diagnose the problem, the systems should have the ability to substitute or isolate services until the ship can make port in an area where depot level maintenance is available.

The primary cost attention for new systems are the unknown factors of the development. The primary reason the military uses COTS components is because they can provide a significant increase in ability while considerably reducing overall development costs. COTS systems do not always provide all the necessary elements required for deployment on military platforms. This is especially appropriate when considering the junction of voice and data. There are no known systems that can control all the known requirements of regulating a data transfer system deploying voice technology or a voice system deploying data technology.

The problem is further compounded because of the lack of cross knowledge between video and data requirements of the developers. It is difficult to find developers with detailed knowledge of both data and voice requirements. There are knowledgeable high level engineers available when developing a system in either technology and there are plenty of third party resources for developing either type of systems, but there are few resources that thoroughly incorporate both specifications. As both systems convert almost exclusively to COTS, the lack engineering resources available to the government for development increases the risk and cost of successful conclusion.

F. MODULAR DESIGN AND OPEN STANDARDS FOR INTEROPABILITY

Modular design is the ability of the system to compartmentalize functionality in such a manner that as new functionality is required or new functionality becomes available the changes needed to upgrade can be implemented with minimal impact on any adjoining functionality. Modular design is achieved by clearly specifying the purpose of the function and by defining the interface at the junction of the function. The actual process within the function does not have to be known and can be developed by independent sources. Implementing a design of this nature creates significant front-end design costs, but provides the best value for the government's overall cost of ownership within life cycle maintenance.

The ability of current telephone systems to maintain open standards for interoperability is minimal at best. The core of all phone systems being installed by the Navy, regardless of the manufacturer, is proprietary in nature. While this method of system design decreases the overall development effort and increases the profits of the manufacturer, which is a necessary part of any product, it severely hampers the ability of the Navy to obtain the best value for its investment dollar.

Developing the system without using open standards at its primary interface junctions hampers the ability to incorporate new technologies. This encumbers the life cycle to the points where it is more cost effect to discard the old system in its entirety and begin anew. By compartmentalizing technologies and providing open standards for interoperability, creating clearly defined points of demarcation where one technology terminates and another begins, gives the system the ability to incorporate new technologies and defers system obsolescence.

Compartmentalizing system functionality decreases the ability to optimize the system performance and increases system design and testing standards requirements. This increase in requirements also significantly increases overall development costs, which is a principal reason why modular design is not utilized. If modular design was applied to the development of the system, the lifecycle support effort would become significantly easier, overall long term support costs could be greatly reduced and life expectancy of the system would be greatly extended.

VI. CONCLUSIONS

A. CONCLUSIONS

The modernization of fleet communications will enhance the Navy's ability to support military operations and how information will be exchanged. The rapid advance in commercial communication is the concept driving network centric warfare. The technology is currently available within the commercial market, but is not yet mature enough to be purchased as a complete, totally encompassed, commercial system.

Rapid advances in technology make it imperative that upgrades be completed within a limited development window. The new system must also evolve to utilize resources not owned, controlled or totally encompassed within the boundaries of the stand alone development. This requires evolving from a tightly controlled centralized process to devices that are loosely coupled where once the information has left the network terminal interface there is limited influence between nodes handling the information.

Attempts at replacing the tactical voice communications system have been a bottom up approach. The developers have started at the lowest level of development with limited regard for the overall objective or interactions of the system. Development needs to be a top down approach. The Navy should develop a comprehensive set of system specifications and a predetermined set of quality of service performance testing standards. Competitive bidding for the actual development of the system will allow the market place to determine the best overall approach and technology for the solution.

The Navy's shipboard tactical communications has extremely unique user and system requirements (Table 19). Just as the differential between software and hardware is becoming less distinguishable, there is a blurring line between technical and political requirements. Technical requirements, as outlined in this document, are only a partial fulfillment of the total requirement specifications. The implementation of the combined tactical voice and data system has to meet several non-software requirements prior to development before it can be considered a viable replacement for the existing systems.

Man power reduction.
Space reduction.
Infrastructure reduction
Incorporate legacy systems.
Maintain QoS at the network terminal.
Retain Total Functionality Of Both Systems

Table 19. Pre-Development Stipulations

There have been several attempts by several Navy agencies to upgrade the tactical voice communications system utilizing commercial components and in-house engineering support to cover the shortfalls. Each of these attempts has ended in failure and the lessons learned have not been realized by the follow on attempts. These efforts did not have a clear understanding of the top level specifications or what criteria will be used in determining the successful completion of the milestones and development.

Although some systems exist in the commercial arena that appear to have most of the qualities and features of the required specifications, a closer examination exposes significant deficiencies. Some services are considering voice over internet protocol (VoIP) as a means to achieve tactical PBX and network convergence. VoIP does have several appealing properties, but also has several inherent limitations that constrain its appeal as a means for tactical communications.

By utilizing well-established standards based quality of service characteristics the Navy can develop a shipboard tactical voice communications system specification. This implementation should incorporate PBX functionality across a distributive data network. By adding gateway components the network will connect the public switched telephone network. The inter-connecting links should incorporate standard PSTN protocols for commercial grade trunking and network terminals.

Communication advances, within and throughout the world, are revolutionizing the industry. In the foreseeable future, massive amounts of information can be instantaneously delivered to any point in the world. Critical information provided in the right place at the right time can reduce the resources required to accomplish the desired objective. The objective of the tactical communication system is to utilize and share common resources among competing systems. However, the flow of information is only providing a limited scope of the requirement; this gets the information from the source to the destination in the most efficient manner while meeting minimum delivery requirements. Future specifications will need to include the ability of the system to compartmentalize the information and present it in a prioritized format.

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